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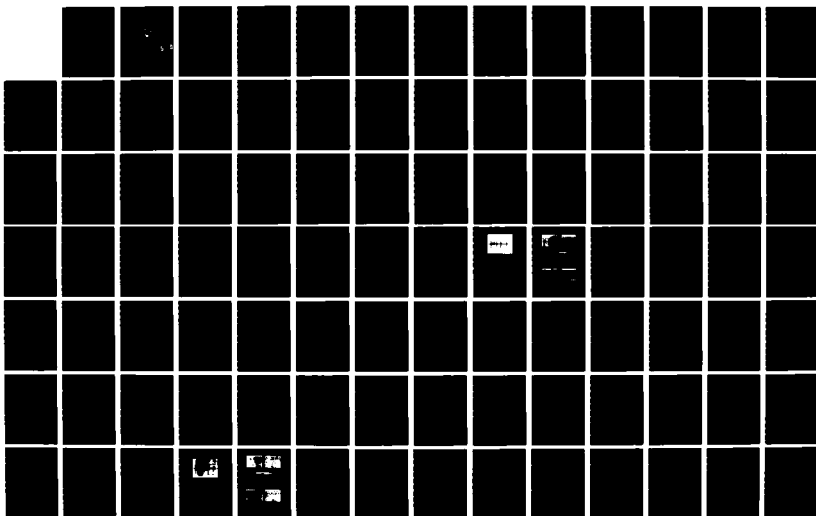
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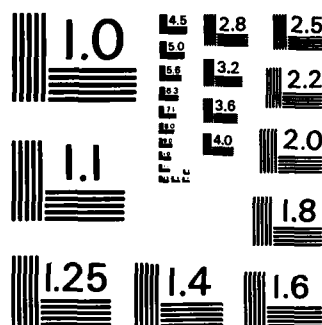
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THESIS

A MATCHED FILTER ALGORITHM FOR
ACOUSTIC SIGNAL DETECTION

by

Dorsett Weston Jordan

June 1985

Thesis Advisor:

R. Panholzer

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**A Matched Filter Algorithm
for Acoustic Signal Detection**

by

**Dorsett Weston Jordan
Lieutenant, United States Navy
B.S., University of Colorado, 1977**

Submitted in partial fulfillment of the
requirements for the degree of

MASTER OF SCIENCE IN ELECTRICAL ENGINEERING

from the

**NAVAL POSTGRADUATE SCHOOL
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ABSTRACT

This thesis is a presentation of several alternative acoustic filter designs which allow Space Shuttle payload experiment initiation prior to launch. This initiation is accomplished independently of any spacecraft services by means of a matched band-pass filter tuned to the acoustic signal characteristic of the Auxiliary Power Unit (APU) which is brought up to operating RPM's approximately five minutes prior to launch.

These alternative designs include an analog filter built around operational amplifiers, a digital IIR design implemented with an INTEL 2920 Signal Processor, and an Adaptive FIR Weiner design. Working prototypes of the first two filters are developed and a discussion of the advantage of the 2920 digital design is presented.

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I. INTRODUCTION

A. BACKGROUND

In April of 1981 a new era of space exploration was opened for mankind when the Space Shuttle successfully returned to Earth after its first orbital mission. Notwithstanding the significant capabilities which this new and radically different concept in space transportation has afforded traditionally large and well funded government and industrial agencies, the Space Shuttle has also ushered in the age of the small experimenter with limited access to funds who has an idea which needs to be tested in space and who now has a means of seeing that test realized.

The Get Away Special (GAS) Program was developed by the National Aeronautics and Space Administration (NASA) to afford qualifying teams of experimenters the opportunity to launch and orbit GAS payloads on a not-to-interfere space available basis during regularly scheduled Shuttle missions.

The main purpose for Shuttle missions is the conveyance into space and deployment of large, sophisticated instruments into Earth orbit. Because many primary mission payloads do not occupy the entirety of the Shuttle payload bay there is frequently opportunity for the additional

deployment of GAS payloads during Shuttle missions [Ref. 1: pp. 8-9].

One significant requirement demanded of GAS experimenters is that each payload be self-contained within its standard size (5.0 cubic foot) NASA provided canister. Thus it is expected that each GAS experiment include provisions within its own confines for all electrical power, heating elements, data control and storage, and so on. In short, GAS payloads may not draw upon any Shuttle services for their normal operation beyond an on-off switch control which may be thrown locally by an astronaut should the Shuttle timeline of events allow for such intervention.

This requirement for a completely self-contained experiment necessitates a well-planned execution of tasks performed under automated control. Although the concession for astronaut involvement in experiment initiation is provided, during some critical periods it is not feasible for astronauts to tend to GAS payload requirements, e. g., during launch and landing phases. At these times it then becomes necessary to design for complete automation of GAS experimental control, to provide most significantly for independent initiation of experimentation.

B. STATEMENT OF GOAL

It is the purpose of this thesis to consider a design algorithm to accomplish an automated initiation of GAS

experimentation based solely upon the passive detection of a well-defined event in the evolution of the Shuttle timeline of operations. Specifically it is desired that an experiment be undertaken to measure the acoustical and vibrational environment present in the Shuttle payload bay from prior to Space Shuttle Main Engine (SSME) ignition (at approximately T-6 seconds) until the Orbiter has exited the Earth's atmosphere (at approximately T+2 minutes).

Previous GAS missions have been deployed to accomplish this same mission goal. In these previous missions the solution to the problem of independent experiment initiation was addressed by way of a simple sound pressure level (SPL) sensor used to detect the unmistakable roar of SSME's as they ignite at T-6 seconds. By this method the sizable impulse thus generated can easily be used to signal experiment initiation. In this way power is applied to data collection and storage elements whose purpose is to record the information relevant to the mission of the experiment.

The obvious shortcoming in this experimental scheme is that it uses as its prime mover the very noise and vibration which it presumes to measure. Though the delay caused by impulse generation and transmission via the SPL sensor may be minimal, it nonetheless requires a finite amount of time for tape transport transients to settle and begin the recording of meaningful data. This delay has been reported to be as much as several seconds and so can be seen to be a

Therefore our task is to develop a matched filter which will detect the presence of a spectral peak at 600 Hz and/or at coincident harmonic multiples of 600 Hz. We will attempt to realize the needed performance emphasizing only the most evident 600 Hz peak. Based upon the results of this design, we will then know if we must consider the additional impact of the less evident harmonic elements.

Our desire is to implement the scheme which provides the simplest, smallest, most reliable, and least power consuming design which will ensure APU detection prior to launch. Therefore detection of all higher order harmonic components is less important than the development of a filter which accomplishes the primary goal. We shall examine both analog and digital methods for implementing our matched filter and compare the effectiveness of the various designs before selecting the device which will orbit the Earth.

but for our purposes this is an inconsequential shortcoming.)

An examination of the PSD's of the remaining sets of data not corrupted by the lack of TSC reveals the expected behavior. In all cases the noise background remains essentially unchanged from the first of the plot pair to the second except for the effect of the impressed APU signature clearly evident in the second. In every plot performed after APU start-up the characteristic 600 Hz peak is evident to some degree and in many instances we also see the harmonic components at 1200 and 1800 Hz. This is especially true for those plots reflecting the environment surrounding microphone 9403 which is located only a bulkhead removed from the APU's themselves. However, of some concern is the fact that the 600 Hz peak is least evident for microphone 9405 which is at the forward bulkhead furthest removed from the APU's and where it is expected that the GAS canister will be placed for the upcoming mission.

The salient points are therefore as follows:

1. The APU does provide a specific signature which becomes clearly evident in the audio spectrum of the Shuttle payload bay pre-launch acoustic environment. There are well-defined spectral peaks at a fundamental frequency of 600 Hz and at integral harmonics thereof.
2. The magnitude of the APU signature is variable in the payload bay depending upon the location of the sensor used to detect it. A sensor placed closer to the after bulkhead will be more apt to respond to the APU signature in a manner which will facilitate matched filter performance, but the signature is evident throughout the payload bay.

it is known that measures were taken to minimize the error introduced in the dubbing process. On most copies a Tape Speed Compensation (TSC) process was employed which ensures that the tape transport travels at the same speed as when originally recorded. That this process is indispensable is realized upon examination of the PSD's obtained for microphones 9405 and 9219 on STS-3. In these two cases alone TSC was not employed due to operator error at a previous generation. The noise floor thus generated is seen to be an order of magnitude greater than in other plots and is severe enough to mask the desired information. In all other tape copies TSC was employed.

(Another generation of tape copies was recorded for use at the Naval Postgraduate School in the development of the experiment described earlier. This recording process was accomplished using a Hewlett-Packard Model 3964A Instrumentation Data Recorder in an FM mode to allow recording of analog data from 0 to 8 kHz. The HP recorder also employs a tape speed servo control which also ensures tape speed accuracy. PSD plots were obtained in all cases for the dubbed tape to verify the accuracy of this latest dubbing routine. In each case the PSD of the dubbed data was confirmed to be a good reproduction of the "original" except for the region below 200 Hz. In this region the reproduction was not as good as in the region above 200 Hz

Appendix B is a summary of PSD plots of the acoustic environment present in the Shuttle payload bay at three different locations for the three missions listed in the previous section. As noted in Figure 2.2 the locations of the three microphones are dispersed throughout the payload bay to reveal variations in the acoustic signature from one location to another. Microphone V08Y9405A (which shall be referred to as 9405 for simplicity) was positioned near the forward bulkhead of the payload bay, furthest removed from the APU. Microphone V08Y9219A (9219) was located low and amidships, and microphone V08Y9403A (9403) was located at the after bulkhead, closest of the three to the APU. All three were identically configured to respond to a dynamic range of 110-157 dB.

The payload bay acoustic data is presented in pairs of plots with the first of the pair showing the signature before the APU is turned on and the second showing the signature approximately thirty seconds later after APU start-up. This data was obtained from an analog magnetic tape copy of the original data recording which was made available from the DATE group at Aerospace Corporation headquarters in Los Angeles.

There is doubtless some extraneous noise introduced in the process by which the original tape was copied to yield the tape available at Aerospace. Furthermore, the generation of the Aerospace copies are not known. However

Table 2.1

DEVELOPMENT FLIGHT INSTRUMENTATION (DFI)
ACOUSTIC MEASUREMENT INFORMATION

Microphone Number	Range (dB)	Cargo Bay Location	Orbiter X	Station Y	Location Z
V08Y9219A	110-157	Internal	863	-100	381
V08Y9220A	110-157	Internal	1190	0	427
V08Y9401A	130-177	External	639	3.5	500
V08Y9402A	130-177	External	1281	4.2	500
V08Y9403A	110-157	Internal	1306	12.0	400
V08Y9404A	130-177	External	1296	0	300
V08Y9405A	110-157	Internal	640	4	423

Note: The frequency response of all microphones is wide-band, 20 Hz to 8 kHz.

D. STS PAYLOAD BAY PRE-LAUNCH ACOUSTIC ENVIRONMENT

As indicated in the previous section we are principally interested in the response of three internal microphones to the Shuttle payload bay environment. The intent is to examine the environment prior to APU start-up to determine the noise background present before the APU acoustic signature is impressed upon this background. Then microphone responses will be examined after APU start-up to reveal the APU acoustic signature over the noise background. Based upon our previous examination of the APU vibrational signature evident in equipment testing we expect to see that spectral peaks at the fundamental and harmonic frequencies of 600 Hz will become clear at the time of APU start-up.

at launch. Rounding out the array of acoustic sensors were three additional microphones mounted externally. All microphones were manufactured by Gulton Industries with only minor alterations differentiating the three external sensors from the four internal ones. The relative location of each of these seven acoustic sensors is shown in Figure 2-2.

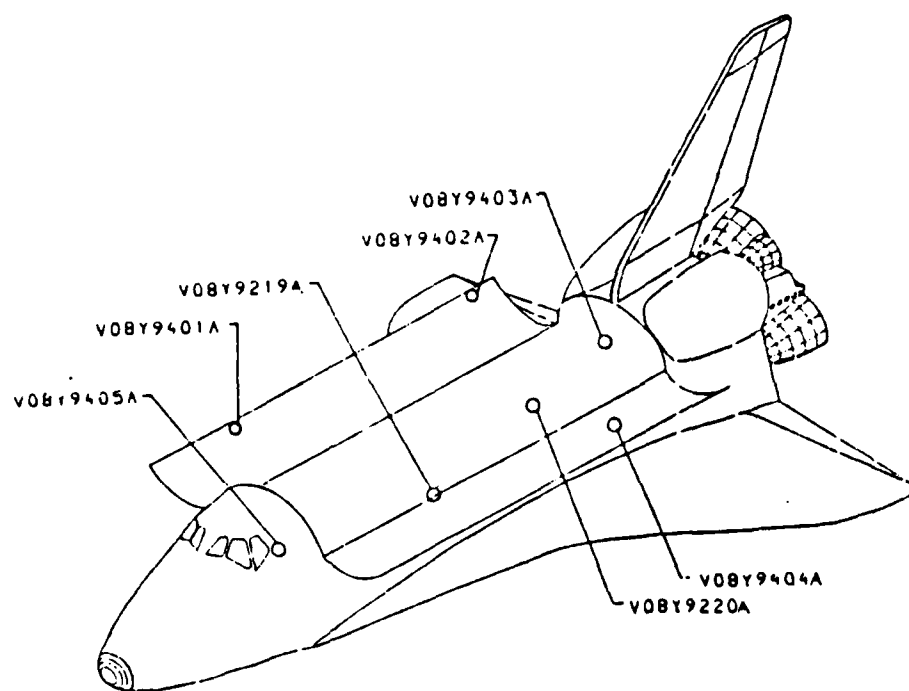


Figure 2.2 DFI Acoustic Measurement Locations

Sensor location designators, dynamic range and frequency response of each microphone are shown in Table 2-1 on the following page [Refs. 3 and 4].

C. STS PAYLOAD BAY ACOUSTIC ENVIRONMENT MEASUREMENT INSTRUMENTATION

Data instrumentation recorders were flown on all early Shuttle flights to measure the acoustic environment present during the launch and landing phases of these missions. These experiments were conducted under the auspices of the NASA DATE (Dynamic, Acoustic and Thermal Environments) Working Group which has as its mission the development of improved methods for predicting all aspects of STS (Shuttle Transportation System) payload environments. Pursuant to this study we shall make use of data collected during each of the following three Shuttle missions:

- STS-2 (conducted November 12-17, 1981)
- STS-3 (conducted March 22-27, 1982)
- STS-4 (conducted June 27 - July 4, 1982)

In each of these missions it is only the data which was recorded just prior to the launch which is of any significance for our purposes here.

Of particular interest to us is the acoustic data recorded at three specific locations in the Shuttle payload bay on each of the three flights listed above. On each flight sixteen selected sensors comprised the STS Development Flight Instrumentation (DFI) system. Of these sixteen sensors, four internal microphones were used to measure the acoustic environment present in the payload bay

of higher spectral components although an expected 2400 Hz peak is present in several plots despite the attenuation.

The significance of these plots is in the consistency of the component spectral elements despite APU loading. Although the relative and absolute magnitudes of the fundamental 600 Hz peak and its harmonic constituents vary somewhat over the range of loading displayed the spectral location of each remains fixed. This shift in magnitudes is of some concern to mechanical engineers as it has been shown that failures in some rotating machinery have occurred when vibration signatures have deviated in this manner. But our concern is with the consistency of the location of the spectral peaks as this confirms the RPM stability of the APU over expected levels of loading.

We thus have a basis for further investigation of the acoustic environment in the Shuttle payload bay. We know that there is a specific vibrational signature which accompanies the normal functioning of the APU and we may expect that this vibration will translate to an acoustic signature in the Shuttle pre-launch environment. We now proceed to investigate this proposition with an examination of the acoustic environment present in the Shuttle payload bay prior to launch.

APU output shaft horsepower is delivered to the hydraulic pump at a nominal 3800 RPM's through a two stage reduction gear. Although balanced to within exceedingly fine tolerances dictated by extremely fast rotational RPM's the APU is nevertheless characterized by a specific vibrational signature. We shall examine this signature in some detail because a careful understanding of its nature is crucial to the development of a filter dedicated to its detection.

In Appendix A there is included a graphical summary of the results of hotfire testing of one APU installed in the Shuttle Orbiter. These are Power Spectral Density (PSD) plots of accelerometers mounted along the x-axis (D0280A) and z-axis (D0281A) of this particular APU for various levels of loading. Also shown for reference is a PSD plot of the background noise prior to APU ignition. Relative to the plots of APU vibration we see that in each case the background noise is no less than two orders of magnitude lower than the PSD peaks of data obtained from the loaded APU.

The results of this test reveal a very particular vibrational signature for the APU. It should be noted that there are consistently repeating peaks at 600 Hz and two harmonics above this value at 1200 Hz and 1800 Hz. The bandwidth of the filter employed in this investigation had a 3 dB rolloff at 2300 Hz. This obviated a close examination

APU MISSION DUTY CYCLE

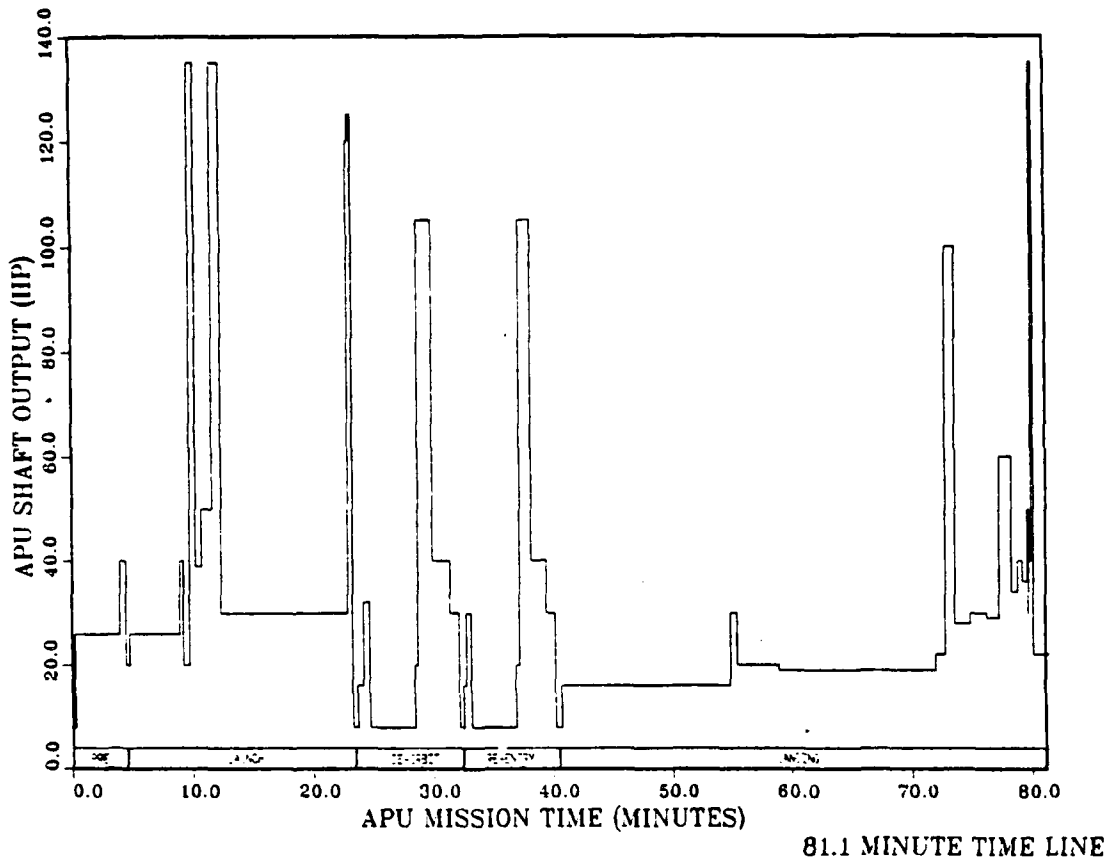


Figure 2.1 APU Mission Duty Cycle
(Baseline 81.1 Minute Time Line)

In the pre-launch phase APU loading varies minimally from 8.0 to 40.0 horsepower according to hydraulic requirements during the phase. Because the APU is designed to operate at 72,000 RPM's (plus or minus eight percent) over its entire range of output shaft horsepower, its steady state and dynamic vibrational characteristics vary little during the pre-launch phase. This is due to the minimal hydraulic loading which characterizes this phase.

II. THE AUXILIARY POWER UNIT (APU) AND THE SHUTTLE PAYLOAD BAY PRE-LAUNCH ACOUSTIC ENVIRONMENT

A. APU DESCRIPTION

The Auxiliary Power Unit (APU) was developed by the Sundstrand Corporation under contract from Rockwell International, the prime contractor for the Space Shuttle. Each Shuttle Orbiter is equipped with three complete APU's and associated hydraulic systems. Each APU and its hydrazine fuel system is independent of the other two during normal operations. However, there are cross-ties between hydraulic systems which allow any two APU's to pick up the load from a third should it fail during operation.

B. APU MISSION DUTY CYCLE

Figure 2.1 is a representative diagram of a typical APU Mission Duty Cycle for an entire Shuttle mission [Ref. 2]. It is expected that a minimum of two restarts from a cold condition will be typical in a mission. The baseline duty cycle calls for 81.1 minutes of APU operation at various power levels from 8.0 horsepower to its maximum rated 135.0 horsepower. This includes launch, de-orbit, re-entry and landing phases and so includes operation at all altitudes corresponding to extremes of airfoil atmospheric resistance and Orbiter speed.

Several different schemes for accomplishing a matched band-pass filter design will be discussed. Nominally this will include an analog design built around operational amplifiers and a digital design of Infinite Impulse Response (IIR) implemented with an INTEL 2920 Signal Processor. This latter configuration will be derived from a cascaded IIR design which results from the Bilinear transformation of the analog filter. The difference equation representation of the digital filter transfer function will form the basis for the 2920 design. In addition I will discuss further design alternatives including an Adaptive Finite Impulse Response (FIR) Wiener filter and an idea for a design centered about a speech processing algorithm. Advantages and disadvantages of each approach will be discussed.

Ultimately one design will be chosen for integration within the GAS experiment just described and scheduled for launch in an upcoming Shuttle mission.

T-5 minutes. These units are essentially jet engines which provide for hydraulic power of Shuttle airfoil control surfaces during the atmospheric phases of launch and landing. They are designed to operate at very high (72,000) RPM's, but also generate a very specific acoustic signature in the audio range of the spectrum during normal operation. If it is possible to detect this APU acoustic signature during the pre-launch sequence of events and to discriminate this signature from among the various other acoustic events which may also occur during the pre-launch phase, then it may be possible to signal experiment initiation by detection of this event.

The emphasis of this thesis will be to describe the nature of the APU acoustic signature and to develop a matched filter which is tuned to its characteristics. It should be emphasized from the beginning that this it is not my intention to develop a classical "matched filter" which rigorously conforms to that definition. I do not have the uncontaminated APU acoustic signature data at my disposal which would allow that sort of an analysis. Rather it is my intention to examine the APU signature in the Shuttle cargo bay environment and to develop a filter which is "matched" to that contaminated signature. It is expected that an extremely narrow (high Q) band-pass filter will accomplish this goal.

significant gap in any serious analysis of the acoustical and vibrational transients which must accompany ignition and launch.

One means of lessening the impact of transient delay is to substitute a solid state data recorder for the traditional magnetic tape variety of instrumentation data recorders. This idea is being investigated by another team of researchers at the Naval Postgraduate School for incorporation into a deployable GAS mission canister. Despite the advantage which a solid state recorder will afford toward minimizing the transient delay prior to meaningful data collection, it can never completely eliminate the transient effect which accompanies any scheme which is trying to measure the same signal which it also uses for experiment initiation. There is a causal imperative here which is inescapable.

It is the purpose of this thesis then to develop a means of experiment initiation which will allow data collection to commence well before SSME ignition. In this manner we will be allowed a full measure of the acoustical and vibrational environment which is present in the Shuttle payload bay during launch.

C. DISCUSSION OF THE GENERAL SOLUTION ALGORITHM

The timeline of Shuttle events prior to launch includes the turn-on of Auxiliary Power Units (APU) at approximately

III. ANALOG FILTER DESIGN THEORY AND IMPLEMENTATION

The traditional method of filter implementation in electronic circuitry was for a long time characterized by an implementation of passive and discrete resistive, inductive and capacitive components tuned to respond to the desired frequency components of the input signal. In the earlier years of circuit design this procedure involved a lengthy process of theoretical development and precise component selection. This was often a tedious process involving component trial and substitution.

Analog filter design took a great leap forward with the advent of integrated operational amplifiers (op-amps) and later, integrated circuit (IC) technology. Now for instance, using an integrated circuit such as the National Semiconductor AF100 Universal Active Filter as a design basis, it is possible for a circuit designer to construct a precise analog filter circuit with a surprising economy of effort. We will use the Biquad Elliptic Filter design which forms the basis of the AF100 to implement the analog designs we shall develop herein.

A. PRACTICAL DEVELOPMENT OF AN ANALOG BAND-PASS FILTER

We will begin our development of a tuned filter design by examining an analog IC implementation of a band-pass

filter with a center frequency of 600 Hz. (We could expand this band coverage to include the two additional center frequencies of 1200 Hz and 1800 Hz if it proves that such a design modification is necessary.) In this development we shall choose to build an Elliptic (or Cauer) filter which exhibits a much steeper roll-off outside the passband over a Butterworth or Chebyshev design of equivalent order. The disadvantage of ripple in the passband, which characterizes Elliptic filters, will cause minimal impact and will not be a factor in the realization of our goal. In fact we can allow the ripple in the passband to be relatively high because our intent is not to pass a faithful representation of the APU signature but only to detect its presence. Thus our goal is to construct a band-pass filter with a high quality factor and narrow passband. This corresponds to a steep roll-off out of the passband.

The method which will be employed to realize this band-pass filter will be to describe the characteristics of the desired analog band-pass design and, using a low-pass-to-band-pass transformation, solve for the form of the corresponding low-pass prototype using transform relations. This will allow us to determine the necessary order of the low-pass design which can subsequently be transformed into the required band-pass filter.

If we again examine the PSD plots of the APU noise above the background for the 600 Hz component we can generalize a

desired filter transfer function to approximate this response. Let us choose the 3 dB points of the band-pass filter (centered at 600 Hz) to be at 575 and 625 Hz. We will allow the pass-band ripple width (PRW) to be as much as 2 dB within the pass-band. Let us furthermore require the stop-band attenuation on either side of the pass-band to be down at least 30 dB at 500 and 700 Hz. This represents a very steep roll-off characteristic and suggests the use of an Elliptic filter design for that reason. In fact, to achieve this degree of roll-off in a low-pass design would require a model prototype of order 6. The equivalent Chebyshev design would require a minimum order of 14, while the Butterworth low-pass filter equivalent order would be at least 63. Clearly the Elliptic design is our only viable alternative.

The low-pass to band-pass transformation for analog filters results in a transfer function which raises the order of the low-pass equivalent by a factor of two. Therefore if we design a band-pass filter it will necessarily consist of a number of second-order stages in the final implementation.

1. Low-Pass to Band-Pass Frequency Transformation

In order to develop the transfer function of an appropriate analog band-pass filter we must begin with the transfer function of the corresponding analog low-pass filter which may be transformed into the desired band-pass

filter by a frequency transformation. However, the characteristics of our final filter are known in band-pass form. Thus we must deduce the analog low-pass design from the band-pass characteristics and then apply the analog transformation to the low-pass prototype to realize our goal. This development is a combination of procedures described in Chen [Ref. 5] and Johnson [Ref. 6].

We wish to design an analog band-pass filter having the following characteristics

$$f'_{p2} = 575 \text{ Hz}$$

$$f'_{q2} = 500 \text{ Hz}$$

$$f'_{p1} = 625 \text{ Hz}$$

$$f'_{q1} = 700 \text{ Hz}$$

$$\text{PRW} = 2.0 \text{ dB (allowable ripple in the passband)}$$

$$\text{MSL} = 30 \text{ dB (minimum attenuation in stop-band)}$$

In this analog development the prime frequencies refer to the band-pass function and the unprimed frequencies to the low-pass function. Figure 3.1 is a graphical depiction of the relationship between the transfer function transform pair. The negative axis frequencies arise from the mathematics of this development. However, we shall only be functionally concerned with the positive axis transformation. We are also considering the low-pass function to be normalized ($\omega = 1$).

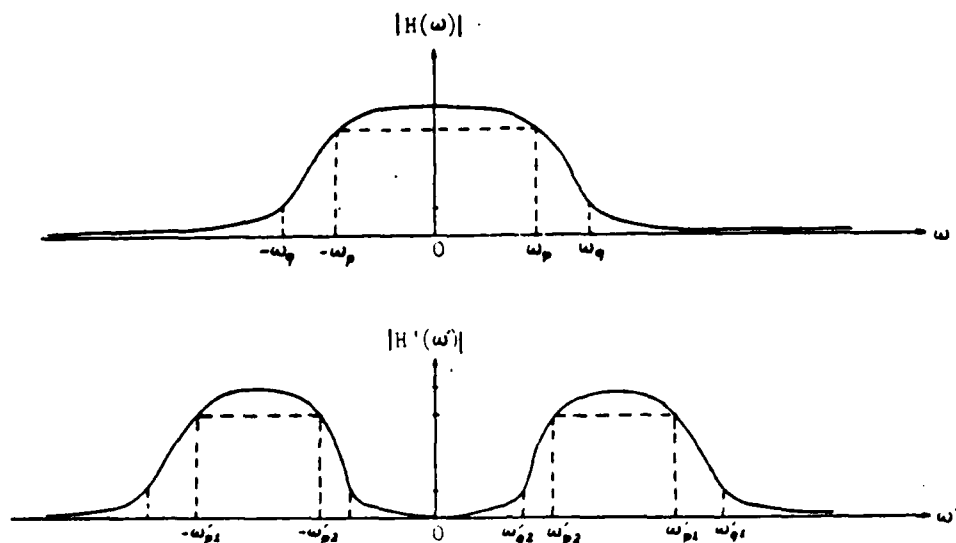


Figure 3.1 A Low-Pass to Band-Pass Frequency Transformation in the Analog Domain

A suitable transformation must therefore accomplish the transform relations detailed in Table 3.1 [Ref. 5].

Table 3.1

LOW-PASS TO BAND-PASS TRANSFORM PAIRS

Low-Pass Function

$$\begin{aligned}\omega &= \infty \\ \omega &= 1 \\ \omega &= -1 \\ \omega &= -\infty\end{aligned}$$

Band-Pass Function

$$\begin{aligned}\omega' &= \infty \\ \omega' &= \omega'_{p1} \\ \omega' &= \omega'_{p2} \\ \omega' &= 0\end{aligned}$$

As developed in Chen [Ref. 5: p. 235], the analog frequency transformation which will accomplish a low-pass to band-pass frequency transformation [$H(s) \Rightarrow H(s')$] is given by the following relation.

$$s = \frac{s'^2 + \omega'_{p1}\omega'_{p2}}{s'(\omega'_{p1} + \omega'_{p2})}$$

or if we substitute the above values

$$s = \frac{s'^2 + 4\pi^2 \cdot (3.59375 \times 10^5)}{s' \cdot 2\pi \cdot 50}$$

or, using $s = j\omega$

$$\omega = - \frac{4\pi^2 \cdot (3.59375 \times 10^5) - \omega'^2}{\omega' \cdot 2\pi \cdot 50}$$

Making the following substitutions into the preceding equation yields

$$\omega'_{q1} = 2\pi \cdot 500$$

$$\Rightarrow \omega_{q1} = -4.3750000$$

and

$$\omega'_{q2} = 2\pi \cdot 700$$

$$\Rightarrow \omega_{q2} = -3.7321429$$

Therefore we are left with the need to design an analog low-pass filter with

$$\omega_c = 1 \text{ rad/sec, and}$$

$$\omega_q = 3.7321429 \text{ rad/sec}$$

This leaves us with a normalized low-pass transition width (TW_{lp}) of

$$\begin{aligned} TW_{lp} &= \omega_q - \omega_c \\ &= 2.7321429 \end{aligned}$$

We now have enough information to enter the tables in Johnson [Ref. 6] to obtain the data shown in Table 3.2.

Table 3.2

ELLIPTIC LOW-PASS FILTER DATA
(for the band-pass filter low-pass prototype)

A	B	C	WZ	WM	KM
21.16400	0.787152	0.842554	4.600435	0.715610	1.258925

2. 600 Hz Elliptic Band-Pass Filter Design

In the case of elliptic band-pass filters the transfer function may be factored into the product of second-order functions. The two factors arising from each second-order low-pass stage have the forms [Ref 6: p. 100]

$$\left[\frac{V_2}{V_1} \right]_1 = \frac{K_1 \sqrt{C/A} (s^2 + A_1 \omega_0^2)}{s^2 + (D\omega_0/E)s + D^2 \omega_0^2} \quad 3.1$$

and

$$\left[\frac{V_2}{V_1} \right]_2 = \frac{K_2 \sqrt{C/A} (s^2 + \omega_0^2/A_1)}{s^2 + (\omega_0/DE)s + \omega_0^2/D^2} \quad 3.2$$

where

$$E = \frac{1}{B} \sqrt{\frac{C + 4Q^2 + \sqrt{(C + 4Q^2)^2 - (2BQ)^2}}{2}}$$

$$D = \frac{1}{2} \left[\frac{BE}{Q} + \sqrt{\frac{(BE)^2}{(Q)^2} - 4} \right]$$

and

$$A_1 = 1 + \frac{1}{2Q^2} (A + \sqrt{A^2 + 4AQ^2})$$

and $Q = f_0/BW = 600/50 = 12$. The coefficients A , B and C are those of the normalized low-pass function given in Table 3.2 above, and K_1 and K_2 are related to the stage gain K by $K = K_1 K_2$.

Equations 3.1 and 3.2 above are of the general form

$$\frac{V_2}{V_1} = \frac{\rho(s^2 + \alpha\omega_0)}{s^2 + \beta\omega_0 s + \gamma\omega_0^2} \quad 3.3$$

which is identical to the form of the low-pass transfer function, except for the replacement of ω_0 by the corresponding low-pass term ω_c .

Our analog band-pass filter will have two stages of the form given by Eq. 3.3. Comparing Eq. 3.3 with Eq. 3.1 and Eq. 3.2 reveals the following transfer function coefficients of the band-pass filter stages [Ref 6: p. 118]:

1) First stage

$$\rho = K_1 \sqrt{C/A}$$

$$\alpha = A_1$$

$$\beta = D/E$$

$$\gamma = D^2$$

2) Second stage

$$\rho = K_2 \sqrt{C/A}$$

$$\alpha = 1/A_1$$

$$\beta = 1/DE$$

$$\gamma = 1/D^2$$

In the FORTRAN program ABPDBP (included as Appendix C) many of the calculations which are indicated in this thesis development will be performed. In Section 1 of this program we begin with desired filter parameters and tabulated values which correspond to the filter we wish to build. We then calculate several derived parameters from these initial values. Next we perform further calculations (which shall be developed shortly) which yield values for filter resistors and capacitors.

In Section 2 of ABPDBP we use the two sets of filter parameters which correspond to each of the filter stages indicated by Eq 3.3 to arrive at the overall transfer function. ABPDBP describes the corrections which must be made to provide for pre-warping of frequencies preparatory to a digital transformation (which shall be discussed in Chapter 4) but these changes can be ignored in this analog discussion. Thus we will use $f_0 = 600$ Hz (which implies the use of ω_0 , not ω_{DIG}) in the program calculations. The fourth order analog filter transfer function which results from this development is given in Equation 3.4.

$$H(s) = \frac{a_4 s^4 + a_3 s^3 + a_2 s^2 + a_1 s + a_0}{b_4 s^4 + b_3 s^3 + b_2 s^2 + b_1 s + b_0} \quad 3.4$$

The values of the coefficients in this analog transfer function representation are given in Table 3.3 which follows.

Table 3.3

ANALOG ELLIPTIC BPF FOURTH ORDER COEFFICIENTS
(Normalized coefficients with gain 0.03981)

Coefficient	Value
a_0	1.0000
a_1	0.0
a_2	2.9587×10^7
a_3	0.0
a_4	1.8991×10^{14}
b_0	1.0000
b_1	2.4351×10^2
b_2	2.7642×10^7
b_3	3.3558×10^9
b_4	1.8991×10^{14}

In Section 2A of the program ABPDBP the poles and zeros of the analog transfer function are then calculated to demonstrate the stability of the filter design. The values for the poles and zeros may be observed in the output of ABPDBP and are reproduced graphically in Figure 3.2 on the following page. The poles of the filter lie within the left

half of the s-plane and this confirms the stability of our design.

3. Analog Band-Pass Filter Simulation

Now that we have developed the transfer function which describes the desired analog band-pass filter we can use this function to simulate the active operation of the

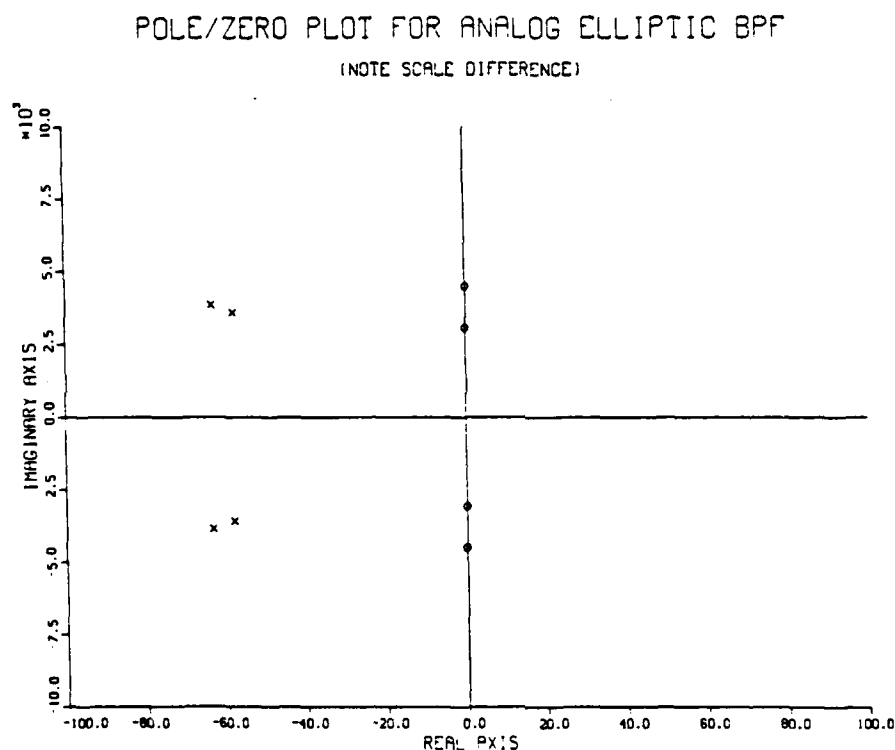


Figure 3.2 Pole/Zero Plot for the Analog Elliptic Band-Pass Filter

filter. The FORTRAN program ABPFR (which is included as Appendix D to this thesis) is used to examine this particular band-pass filter simulation. Figures 3.3, 3.4

and 3.5 which follow are the results of this computer simulation of the filter response for the device we have just designed. The range of frequencies of the simulated computer input is DC to 1 kHz. The simulated amplitude is constant over the range of input frequencies.

In Figure 3.3 we see the amplitude response which is near zero at all but the passband frequencies around 600 Hz. Between 500 and 700 Hz we confirm the desired filter response. The center frequency is located at 600 Hz and

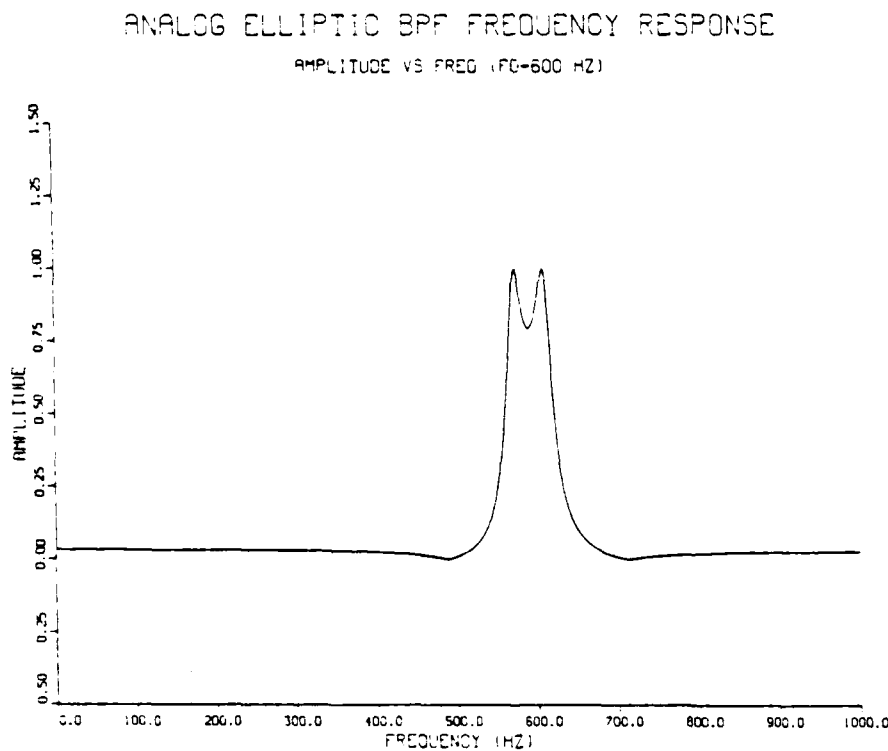


Figure 3.3 Analog Elliptic BPF Frequency Response
(Computer Simulated Amplitude Response)

there is a significant minimum at the center frequency due to the effect of the passband ripple of 2.0 dB. Furthermore we observe a half power point (3.0 dB down point) at about 575 Hz and 625 Hz as specified in our design.

In Figure 3.4 we again observe a computer simulation of the amplitude response of the filter, this time measured in decibels. The marked presence of notches at about 500 Hz and 700 Hz is obvious, and the 30 dB minimum loss in the stopband is also confirmed. We also have graphical confirmation of the 2.0 dB ripple width in the passband.

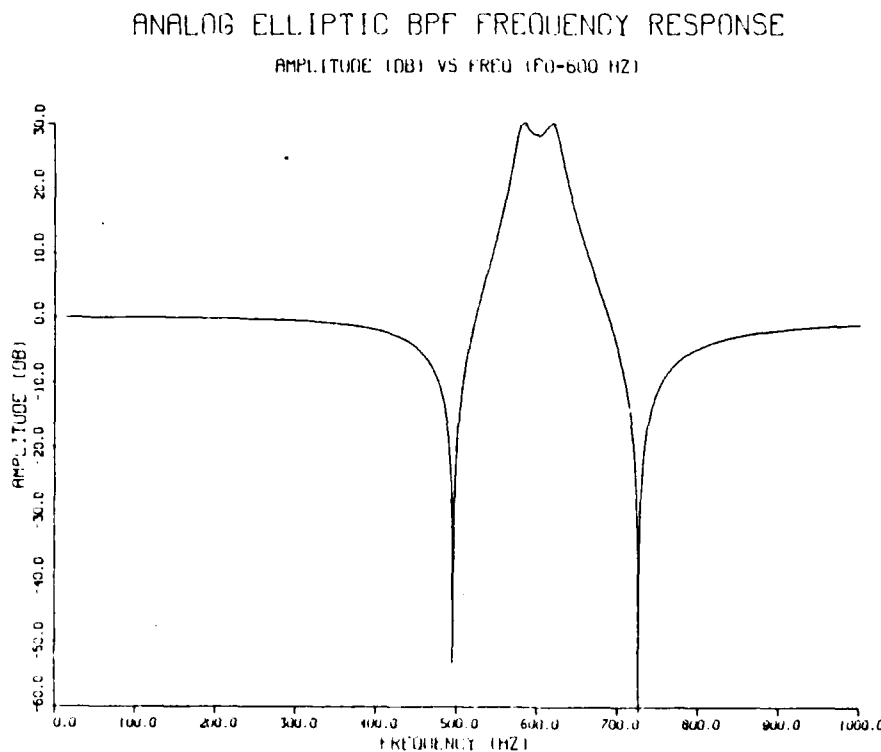


Figure 3.4 Analog Elliptic BPF Frequency Response
(Computer Simulated Amplitude Response in dB)

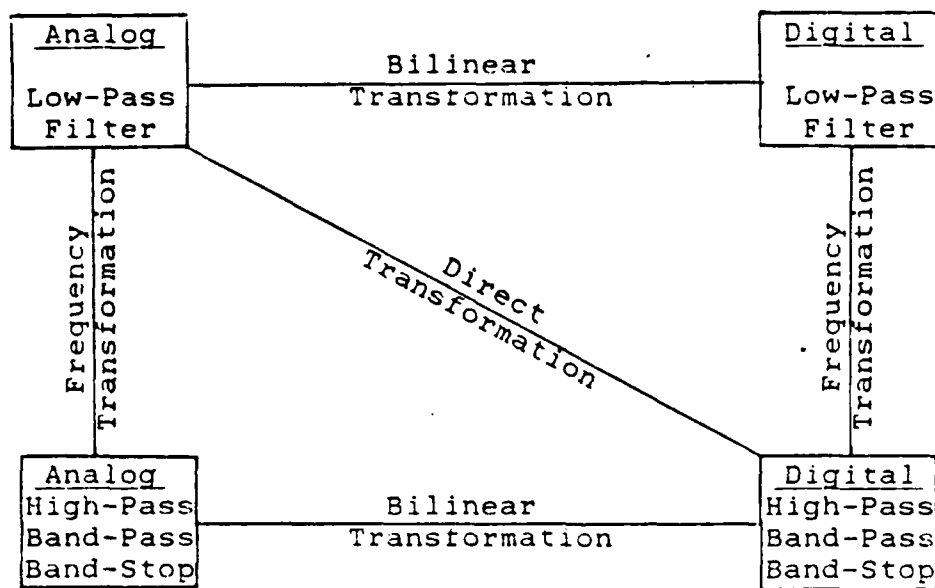


Figure 4.1 Analog and Digital Frequency Transformations

It should be reiterated that our goal in this section will be to develop a digital filter of Infinite Impulse Response (IIR) characteristics. This means that our filter will use the results of previous outputs to realize a later output. Although Finite Impulse Response (FIR) digital filters offer several qualitative advantages over IIR designs in the areas of phase linearity, stability, and an inherent protection against round-off error, they also require a larger number of delay elements to realize a design with a steep filter roll-off. This will be of concern to us when we realize an implementation in hardware with devices limited to a relative few number of filter transfer function poles and zeros.

case known analog filter characteristics in the frequency domain (the Laplacian "s" domain) are converted to similar characteristics in the digital "z" domain. Each of these techniques introduces a non-linearity into the resulting amplitude and phase characteristics of the original analog filter. If necessary to preserve the phase, equalizers may be employed to return the phase characteristic to a nearly linear behavior over the region of interest in the digital domain. In our case any phase distortion can be ignored because we are only interested in frequency detection and not accurate reproduction.

Generally speaking, when beginning with an analog low-pass design, we may proceed in a number of ways to arrive at a corresponding digital band-pass filter realization. For instance, we may first transform the low-pass filter to an analog band-pass design (as we did in Chapter 3 for the low-pass to band-pass transformation) and then employ an analog to digital transformation to yield the digital filter. Alternatively we may choose to employ the analog to digital transform on the low-pass filter and then apply a digital low-pass to digital band-pass transformation to realize our goal. Finally, it is also possible to combine these two-step routines into a single-step analog low-pass to digital band-pass direct transformation. These options are shown in Figure 4.1 [Ref. 5: p. 269].

IV. DIGITAL FILTER DESIGN

When designing an IIR digital filter for a specific application it is common practice to first develop an analog filter with appropriate characteristics as we did in the preceding chapter. Once the analog design is attained it is then possible to transform this analog filter into a digital filter with the desired passband characteristics.

There are several reasons why it is desirable to use this approach [Ref. 7: p. 5-7]. Of primary importance is the fact that the art of analog filter design is highly advanced. Consequently there are many techniques available for implementing specific designs. Because useful results can be achieved, following established analog design procedures presents advantages in the amount of effort which must be spent in the design phase.

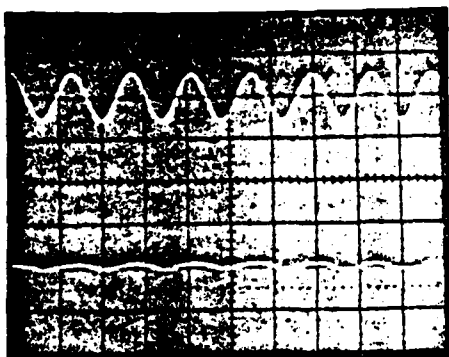
Additionally, many useful analog design methods have relatively simple closed-form design formulas. This greatly facilitates the implementation of the corresponding digital filters.

Finally, in many applications it is of interest to use a digital filter to simulate the performance of an analog linear time-invariant filter.

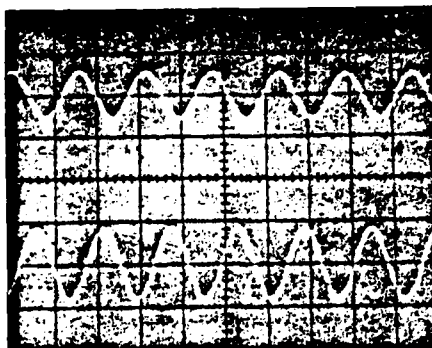
There are many alternative methods for accomplishing a transformation of fixed filter characteristics. In each

In Figure 3.10 we examine this response more specifically for discrete frequencies in the range of 500 Hz to 700 Hz. Instead of applying a ramped sinusoid we input five discrete sinusoids while maintaining a constant amplitude. Thus we again observe the very narrow bandpass filter response at least within the limits presented here. We also confirm the rapid shift in phase of 180 degrees from the lower to the upper bound in agreement with theory. While this examination by itself does not confirm the desired filter response, it does so when considered with the results of the previous figure.

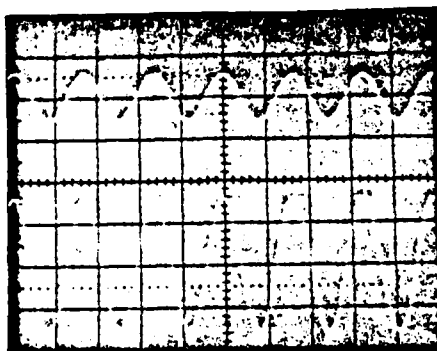
In Chapter 4 we will use the results of this analog filter implementation to develop an equivalent digital realization. To do this we will use common transformation techniques to arrive at a z-domain transfer function which we will then reduce to a difference equation. This format will then allow us to realize a digital elliptic filter by use of the INTEL 2920 Signal Processor. This hardware realization of the digital filter will be accomplished in Chapter 5.



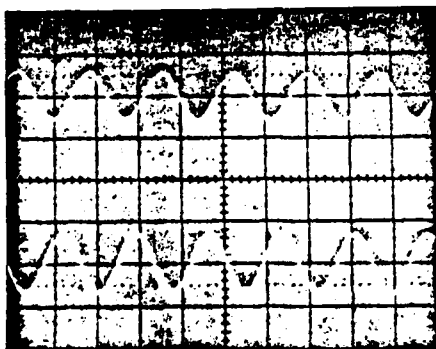
a) 500 Hz



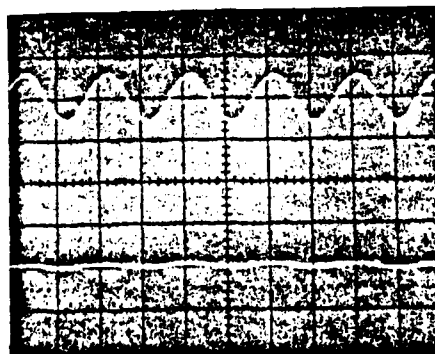
b) 575 Hz



c) 600 Hz



a) 625 Hz



b) 700 Hz

Figure 3.10 Analog Elliptic BPF Frequency Response
 Upper trace (Input): 50 mV/div scale
 Lower trace (Output): 1.0 V/div scale

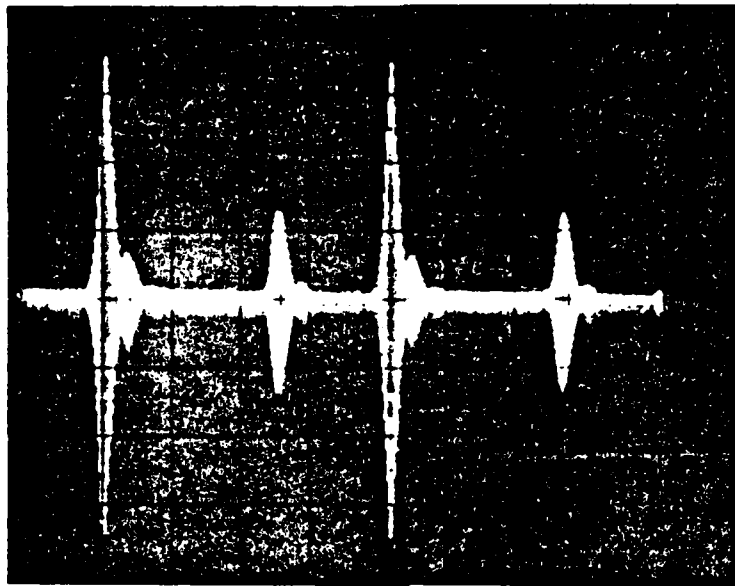


Figure 3.9 Analog Elliptic BPF Frequency Response
(Photograph of Actual Filter Amplitude Response
to a Ramped Sinusoidal Input)

This gives rise to the appearance of a double response which is noted in the figure. Actually we are observing a multiple response over successive up and (faster) down ramps of the input sinusoid. Thus we are able to observe graphical confirmation of the filter amplitude response predicted in the foregoing discussion.

As expected we observe a very narrow filter bandpass response (with frequency limits we will look at more closely in the following paragraph). The curious extended response ("hump") at the upper end of the passband is due to the inexact placement of poles and zeroes accomplished by tuning of the filter response in the aforementioned manner.

The two 74161 counter stages which follow the multivibrator are designed to count up to 255 occasions of the threshold being exceeded in a .5 second period before the decision is made that a valid 600 Hz signal was detected. This is an arbitrary figure. The .5 second period is established by the 555 timer which is also fed by the one-shot multivibrator. If the counter stages do not sum to 255 within a .5 second period then the 555 resets the counter stages to zero and counting begins anew with the comparator. If the counter does reach 255 within the .5 second period then a latch is set for the remainder of the .5 second period. This TTL level signal is the one which provides the microprocessor interrupt indicating that the APU signal has been detected.

C. ANALOG BAND-PASS FILTER IMPLEMENTATION RESULTS

Figure 3.9 on the following page is a photograph of the actual frequency response of the analog elliptic band-pass filter we have developed. A sweep generator was used to input a ramped sinusoidal input comprised of a linear continuum of frequencies (generated by application of a skewed triangular input to a voltage controlled frequency oscillator) in the range of approximately 100-1000 Hz. Because the ramp generator does not exhibit an instantaneous return, the return also generates a down-frequency response albeit at a rate greater than that of the up-frequency ramp.

which follow the amplifier. If the amplitude of the amplified filter output goes above the threshold set at the reference input of the LM311-based comparator then a pulse is developed for the duration that the input signal exceeds the threshold level. A negative-edge triggered 74121-based one-shot multivibrator follows the comparator. It is designed to send a one millisecond pulse to the counter stages which follow any time the comparator detects an input signal which exceeds the threshold level.

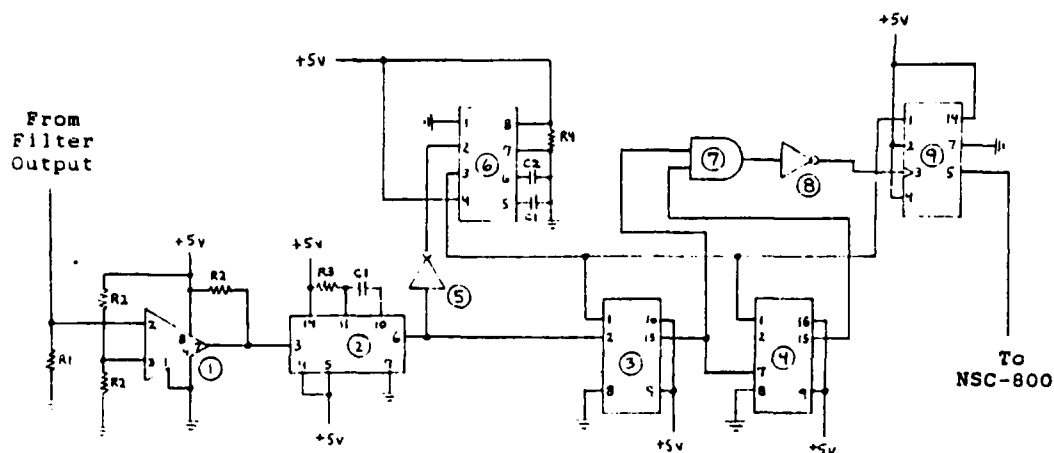


Figure 3.8 Follow-Up Pulse-Shaping Logic Circuitry

Microcircuits

- 1: LM311 Op-Amp
- 2: 74121 1 msec One-Shot
- 3: 74161 4 Stage Counter
- 4: 74161 4 Stage Counter
- 5: 7404 Inverter
- 6: 555 0.5 sec Timer
- 7: 7432 AND Gate
- 8: 7404 Inverter
- 9: 7474 D-Type Flip-Flop

Components

- R1: 100 k Ω
- R2: 20 k Ω
- R3: 15 k Ω
- R4: 470 k Ω
- C1: 0.1 μ F
- C2: 1.0 μ F

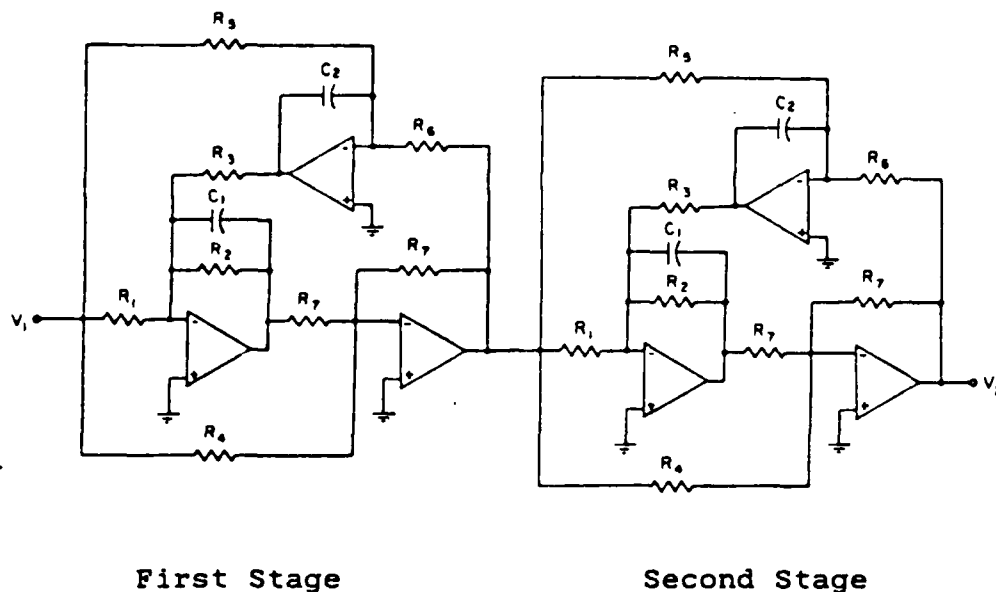


Figure 3.7 Biquad Elliptic Band-Pass Filter Circuit

sinusoidal nature and inadequate to drive a microprocessor interrupt designed to accommodate TTL logic levels. Thus we must include further circuitry into our design which will send a TTL compatible logical signal to the microprocessor when the APU signal is detected. The circuit which accomplishes this is shown in Figure 3.8.

The output from the filter stages is first sent to a linear amplifier constructed around a 741 op-amp. Because the APU signal is low voltage out of the microphone detector it is necessary to amplify the filter output prior to logical evaluation.

The decision of whether or not a 600 Hz component is present is the function of comparator and counter elements

possible to these values and then tuning the circuit for the desired performance. Tuning is accomplished in each stage by adjusting R_4 to set the notch frequency f_z , R_3 to set the center frequency f_0 , R_2 to set Q , and R_1 or R_5 to set the gain.

Table 3.4

BIQUAD ELLIPTIC BAND-PASS FILTER COMPONENT VALUES

1st Stage		2nd Stage	
Component	Value	Component	Value
C_1	.00996	C_1	.01030
C_2	.00995	C_2	.01034
R_7	26.7	R_7	26.7
R_6	25.7	R_6	26.5
R_5	10.5	R_5	20.2
R_4	14.8	R_4	14.8
R_3	25.7	R_3	26.6
R_2	785.	R_2	813.
R_1	437.	R_1	452.

Note: Capacitor values are μF , resistor values are $k\Omega$.

The resulting schematic for the fourth-order Biquad elliptic band-pass filter is shown in Figure 3.7.

2. Follow-Up Logic Circuitry

When the band-pass filter is implemented the effect is to produce a response which narrowly limits the passband to within a few tens of hertz about the center frequency of 600 Hz. Still, the output of this filter will be of

Similarly, the second stage values are given by

$$R_1 = \frac{DE\sqrt{A/C}}{K\omega_0 C_1}$$

$$R_2 = KR_1\sqrt{C/A}$$

$$R_3 = \frac{D}{\omega_0 C_1}$$

$$R_4 = \frac{A}{C}\sqrt{R_1/K}$$

$$R_5 = \frac{A_1\sqrt{A/C}}{KD\omega_0 C_2}$$

$$R_6 = \frac{C_1 R_3}{C_2}$$

In Section 1 of the program ABPDBP (introduced as Appendix C) we calculate the resistor values for the Biquad band-pass circuit just discussed. The resulting component values which were thus derived are shown in the appendix and are also included here as Table 3.4. These are computed values for resistance and capacitance. In fact the circuit is constructed by selecting standard values as close as

coefficients and the derived complements we have already evaluated. We begin our development by choosing a standard value for C_1 (given roughly by $10/f$, μF) and then proceeding to calculate elemental values. In the equations which follow the values for C_1 and R_1 are arbitrary within limits, and are chosen to minimize the spread of resistance values. We pick $C_2 = C_1$ and $R_1 \approx 1/(\omega_0 C_1)$. A_1 , E and D are as given previously.

The first stage values are thus [Ref. 6: p. 126]:

$$R_1 = \frac{E\sqrt{A/C}}{KD\omega_0 C_1}$$

$$R_2 = KR_1\sqrt{C/A}$$

$$R_3 = \frac{1}{D\omega_0 C_1}$$

$$R_4 = \frac{A}{C}\sqrt{R_1/K}$$

$$R_5 = \frac{D\sqrt{A/C}}{KA_1\omega_0 C_2}$$

$$R_6 = \frac{C_1 R_3}{C_2}$$

B. HARDWARE IMPLEMENTATION

1. Biquad Analog Band-Pass Elliptic Filter

There are many ways to perform a hardware implementation of the analog band-pass transfer function we have just developed. One relatively easy method employs the use of op-amps as the active filter component. We shall use a Biquad op-amp filter implementation which exhibits good stability and ease of tuning. Additionally, implementation is made simpler by the use of a 74124 quad op-amp microchip which allows a single chip per second-order stage. The generalized circuit diagram for a second-order stage of a Biquad filter is shown in Figure 3.6 [Ref. 6: p. 127].

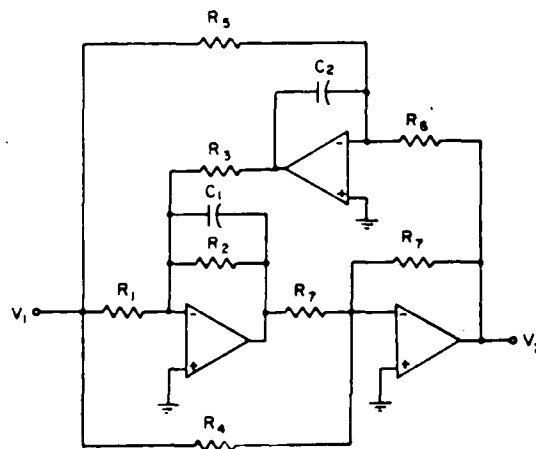


Figure 3.6 Biquad Elliptic Filter Circuit

Component resistor and capacitor values for the Biquad filter depend upon the low-pass normalized

Finally, in Figure 3.5 we view the computer simulation of the analog filter phase response. Although our application is not phase dependent (due to the fact that it is the presence alone of the 600 Hz element which is of concern to our circuit--not its accurate transmission), we do confirm the significant effect upon phase for our elliptic filter in the passband between 500 Hz and 700 Hz. Between these two frequencies we observe a 360 degree shift in phase.

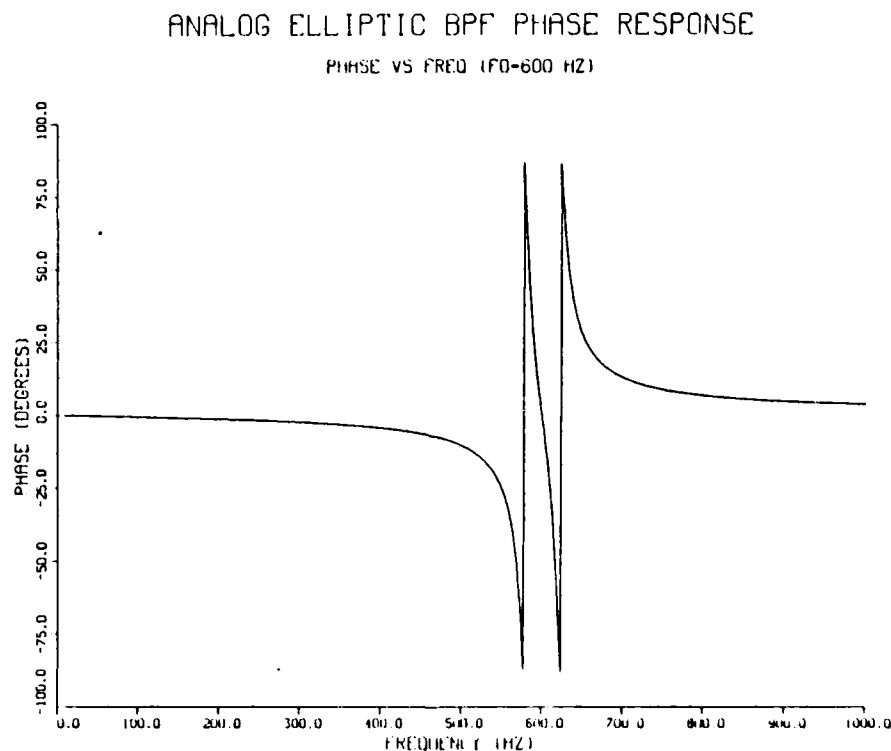


Figure 3.5 Analog Elliptic BPF Frequency Response
(Computer Simulated Phase Response)

A. ANALOG BAND-PASS TO DIGITAL BAND-PASS FILTER DESIGN

An analog-to-digital bilinear transformation makes it possible to apply a relation which transforms an analog band-pass filter into a desired band-pass digital design.

We will realize the 600 Hz digital bandpass filter by applying the bilinear transformation to the transfer function of the analog band-pass filter we developed in Chapter 3. Once implemented we will examine the performance of this filter in view of our goal of APU start-up identification. If necessary we will determine which refinements and modifications to our design may be necessary to realize our goal.

We recall from Chapter 3 that the transfer function of the analog elliptic band-pass filter was given by the product of the two second-order functions given by Eqs. 3.1 and 3.2. This product has been computed (as shown in Appendix C) and is found to be

$$\frac{V_1}{V_2} = \frac{\rho(s^4 + Fs^3 + Gs^2 + Hs + J)}{s^4 + Ms^3 + Ns^2 + Ps + Q}$$

where

$$\rho = 0.039811$$

$$F = 0$$

$$G = 29.587 \times 10^6$$

$$H = 0$$

$$J = 189.91 \times 10^{12}$$

$$M = 0.24351 \times 10^3$$

$$N = 27.642 \times 10^6$$

$$P = 3.3558 \times 10^9$$

$$Q = 189.91 \times 10^{12}$$

The poles of the analog band-pass filter transfer function were shown (graphically in Figure 3.5) to be:

$$-63.567 \pm 3.8421j \times 10^3$$

$$-58.188 \pm 3.5859j \times 10^3$$

The analog filter is therefore stable.

To transform the analog band-pass transfer function into a digital version we will use the Bilinear Transformation [Ref. 8: pp. 219-224] which is characterized by the following relation

$$s = \frac{2}{T} \frac{z - 1}{z + 1} \quad 4.1$$

This transformation will map stable analog poles which are in the left-half of the s-plane into the interior of the unit circle in the z-plane. Thus stability is preserved in all cases.

If we then make the substitution $s = j\bar{\omega}$ and $z = e^{j\omega T}$ into Eq. 4.1 and simplify, we can establish the relationship between the frequencies in the analog and digital cases. (In this and the following discussion we shall denote frequencies in the analog case with an overbar ($\bar{\omega}$), and those in the digital case without one (ω).)

The resulting relation is

$$\bar{\omega} = \frac{2}{T} \tan \frac{\omega T}{2} \quad 4.2$$

where T is the sample period given by $1/f_s$. In this case $f_s = (6.666 \times 10^6)/(4 \times 192)$, which we will show shortly. This results in $T = 1.15212 \times 10^{-4}$.

This relationship between analog and digital frequencies is shown in Figure 4.2 and reveals that the Bilinear Transform does not provide a linear mapping from one function to another. The frequency range from 0 to ∞ in the continuous case is warped into the frequency range from 0 to π/T in the digital case.

Therefore, if we have an analog filter with transfer function $H(s)$, we may then perform the following substitution dictated by Eq. 4.1

$$H(z) = \bar{H}(s) \big|_{s=(2/T)[(z-1)/(z+1)]} \quad 4.3$$

Another way of expressing this same relation is

$$H(e^{j\omega T}) = \bar{H}(j\bar{\omega}) \mid \bar{\omega} = (2/T) \tan \omega T/2$$

Using this relation the characteristics of $H(z)$ can be obtained graphically from those of $H(s)$ as shown in Figure 4.2 [Ref. 5: p. 262].

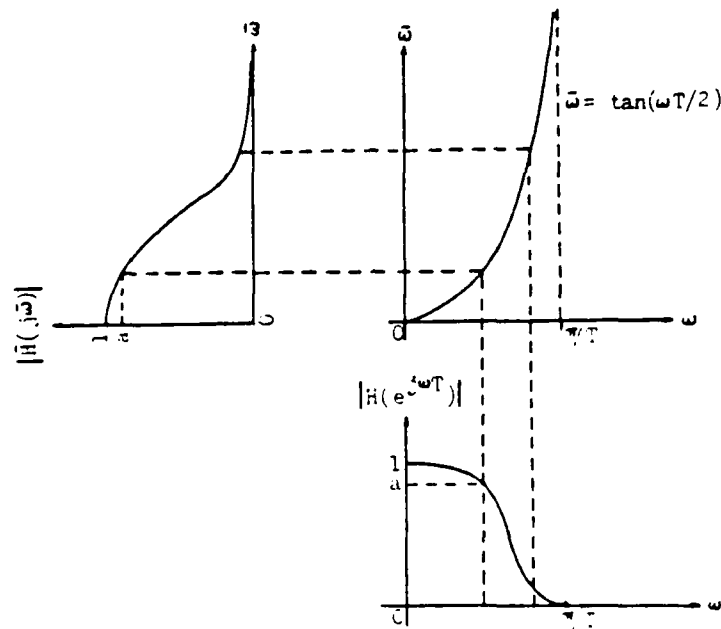


Figure 4.2 The Bilinear Transformation (showing analog and digital transfer functions and the non-linear warping of frequencies.)

We see from the figure that there is no aliasing problem associated with the transform because the frequency is limited to less than π/T (8680π) in the digital case. However, because of the frequency warping we have to make a proper transform of frequency according to Eq. 4.2

before application of the Bilinear Transform. Consequently, for a transformation of the analog band-pass filter derived as Eq. 3.3, we must substitute $f_c = 590.825$ Hz for $f_c = 600$ Hz before application of the Bilinear Transform to ensure a proper transformation to the digital domain. Once this is accomplished all we need do is apply the Bilinear Transform to the resulting "pre-warped" analog band-pass filter transfer function.

The FORTRAN based computer program previously introduced in Appendix C also provides for this development and implements Eqs. 4.2 and 4.3 to derive the following digital transfer function $H(z^{-1})$ for the desired digital band-pass filter

$$H(z^{-1}) = \frac{\rho(1 + Fz^{-1} + Gz^{-2} + Hz^{-3} + Jz^{-4})}{1 + Mz^{-1} + Nz^{-2} + Pz^{-3} + Qz^{-4}} \quad 4.4$$

where the constants are as follows

$$\rho = 0.039516$$

$$F = -3.6279$$

$$G = 5.2861$$

$$H = -3.6279$$

$$J = 1.0000$$

$$M = -3.6251$$

$$N = 5.2586$$

$$P = -3.5768$$

$$Q = 0.97353$$

The poles of the transfer function given by Eq. 4.4 are

$$0.90781 \pm 0.40813j \quad (\text{Magnitude} = .99533)$$

$$0.90475 \pm 0.40493j \quad (\text{Magnitude} = .99123)$$

POLE/ZERO PLOT FOR DIGITAL BPF

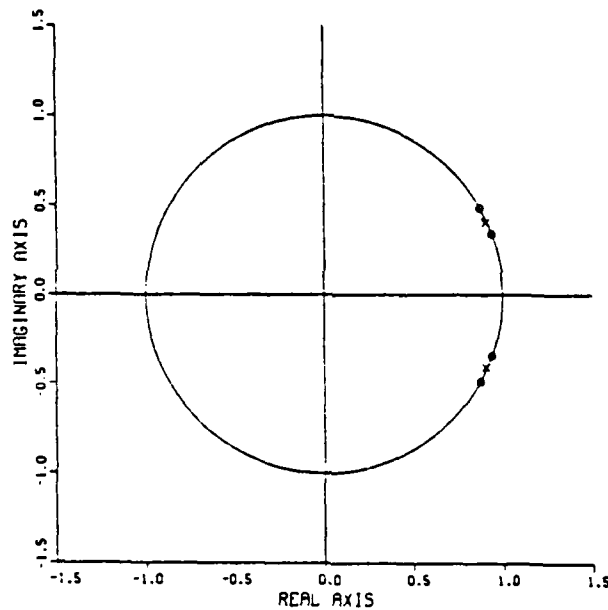


Figure 4.3 Pole/Zero Plot for the Digital Elliptic Band-Pass Filter (poles appear singular, but are in fact double and nearly coincident)

and thus we confirm the mapping of stable poles in the analog domain into stable digital poles located inside the

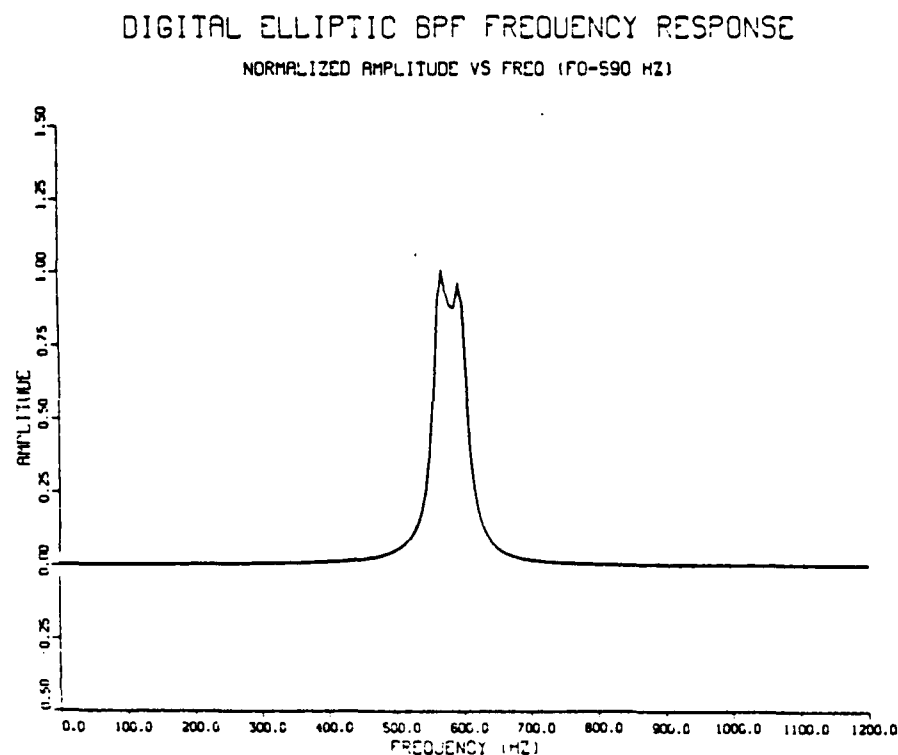
unit circle. This digital pole/zero plot is shown graphically in Figure 4.3 on the previous page.

B. DIGITAL BAND-PASS FILTER SIMULATION

Equation 4.4 represents the digital filter transfer function equivalent to the analog filter transfer function we presented in Chapter 3. In a manner completely analogous to that development we are now able to demonstrate a computer graphical simulation of the digital filter frequency and phase response and compare these to the previous results. The FORTRAN program used to present this graphical output is included in Appendix E under the title DBPFR.

In Figure 4.4 we see the digital filter frequency response and observe that it is nearly identical to the analog response in consonance with our design goal. The minor differences are remarkable and explicable. The center frequency of the digital filter is diminished to the pre-warped center frequency of approximately 591 Hz. Additionally, the two peaks of the amplitude response located at about 585 Hz and 595 Hz are not of equal magnitude. This is due to the difference of pole proximity to the unit circle. Although the poles appear coincidental in the graphical presentation in Figure 4.3, they are actually distinct; the pole nearer to the real axis is some

.004 units closer to the unit circle which accounts for the amplitude disparity between the two poles.



**Figure 4.4 Digital Elliptic BPF Frequency Response
(Computer Simulated Amplitude Response)**

In Figure 4.5 we observe the digital filter frequency response as measured in dB. This curve appears somewhat different from its analog counterpart but the important feature is maintained. A steep filter rolloff is realized out of the passband and the response is diminished by about 30 dB at approximately 500 Hz and 700 Hz according to design specifications. Although the analog filter did not deviate much from this 30 dB down figure we see an added benefit of

the digital filter wherein the rolloff continues monotonically over our observed spectrum.

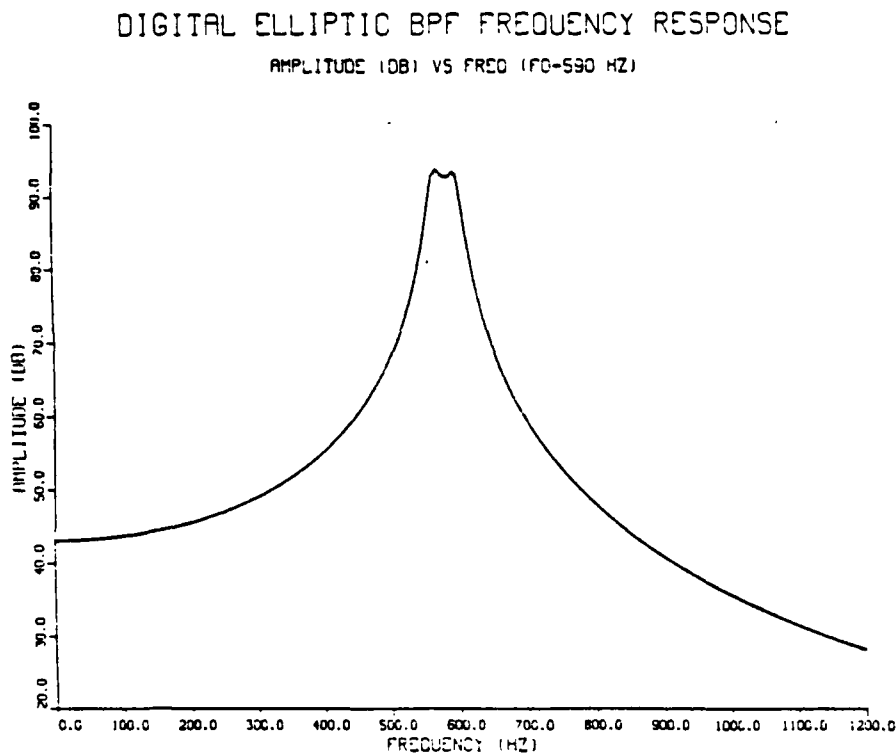


Figure 4.5 Digital Elliptic BPF Frequency Response
(Computer Simulated Amplitude Response in dB)

Finally, in Figure 4.6 we observe the phase response of our digital filter. Once again this closely approximates the severe phase distortion we observed with the analog filter although the center frequency is again confirmed to be significantly less than the nominal 600 Hz we expected of the earlier filter design. To reiterate, this phase distortion is a hallmark of elliptic filters and the

Bilinear transformation, but our application is not phase dependent. Thus we may ignore this effect.

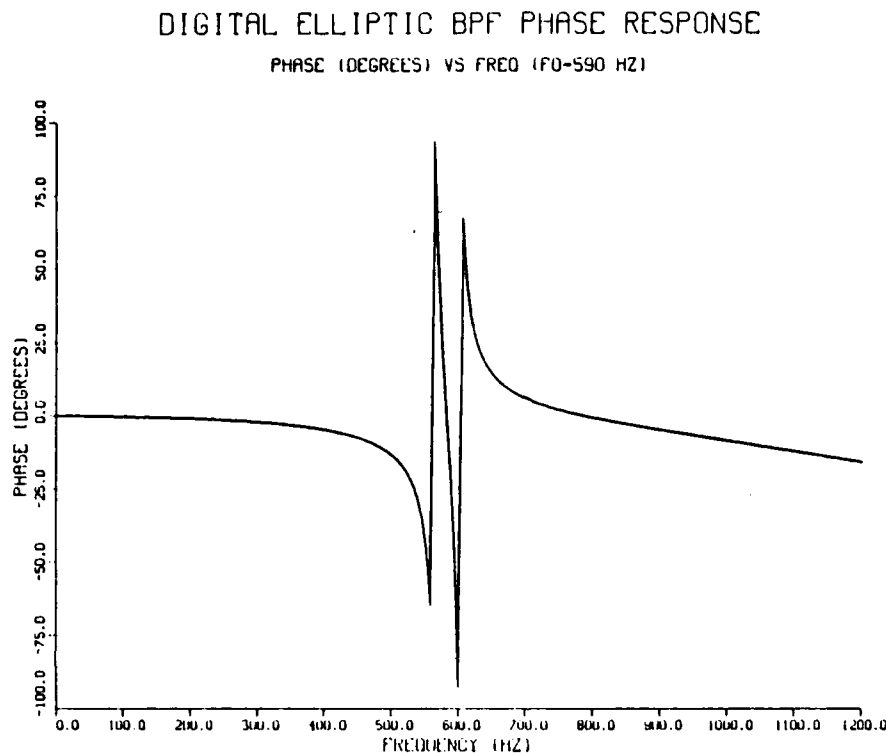


Figure 4.6 Digital Elliptic BPF Frequency Response
(Computer Simulated Phase Response)

C. DIFFERENCE EQUATION REPRESENTATION

The transfer function for the fourth order Elliptic Filter was given previously in Equation 4.4. Section 4 of the program ABPDBP introduced earlier in Appendix C accomplishes a transformation of this quotient of fourth order polynomials and provides an equivalent cascaded

representation of two second order filter stage blocks, each of the form

$$H(z) = \frac{Y(z)}{X(z)} = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2}}{b_0 + b_1 z^{-1} + b_2 z^{-2}}$$

where $X(z)$ and $Y(z)$ refer to the filter input and output, respectively. The values of the dual quadratic coefficients of the cascaded second order transfer function and as computed by the program in Appendix 4 are given in Table 4.1.

Table 4.1

ELLIPTIC BPF SECOND ORDER STAGE COEFFICIENTS

1st Stage		2nd Stage	
Coefficient	Value	Coefficient	Value
a_0	1.0000	a_0	1.0000
a_1	-1.7503	a_1	-1.8780
a_2	1.0017	a_2	0.9981
b_0	1.0000	b_0	1.0000
b_1	-1.8163	b_1	-1.8092
b_2	0.9910	b_2	0.9822

Both second order z -domain filter stage transfer functions can be manipulated in a familiar way to realize the following z -domain difference equation.

$$b_0 Y(z) = a_0 X(z) + a_1 X(z)z^{-1} + a_2 X(z)z^{-2} \\ - b_1 Y(z)z^{-1} - b_2 Y(z)z^{-2}$$

Applying the inverse z-transform to this z-domain difference equation yields the time domain digital difference equation .

$$b_0 y(k) = a_0 x(k) + a_1 x(k-1) + a_2 x(k-2) \\ - b_1 y(k-1) - b_2 y(k-2)$$

4.5

The Signal Flow Graph corresponding to the difference equation given by Eq. 4.5 is shown in Figure 4.7. The difference equation representation is important because this is the basis for the hardware implementation of the digital filter which shall follow in Chapter 5.

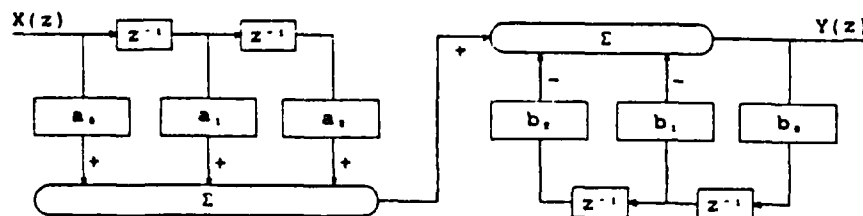


Figure 4.7 Second Order Stage Signal Flow Graph

D. DIFFERENCE EQUATION IMPLEMENTATION SIMULATION

We have just stated that the difference equation representation of the digital filter will become the basis for the INTEL 2920 Signal Processor implementation which is to follow. In order to show the adequacy of this method of implementation it is useful to demonstrate the impulse response and filter frequency response by computer simulation. In the following chapter sections we will demonstrate these simulations and show that they yield results in keeping with our design expectations.

1. Digital Filter Impulse Response

In Appendix F is presented the complementary FORTRAN programs S22I and S22IG. Both implement the impulse response simulation for the difference equation representation of the digital elliptic band-pass filter. In the case of S22I the output is digital and is shown to exemplify the filter response over a greater period of time than is usefully represented otherwise. In the case of S22IG the output is graphical and will be presented here.

In both these programs the impulse is equal to the greatest allowable input which guarantees an output of less than unity. This is done for reasons of filter stability as well as a limitation of the INTEL 2920 which will be discussed in the next chapter. By examining the impulse response we confirm the stability of the filter design by ensuring that the output decays to zero over time. In

addition we can observe the natural response of the system by establishing the frequency of the decaying sinusoid.

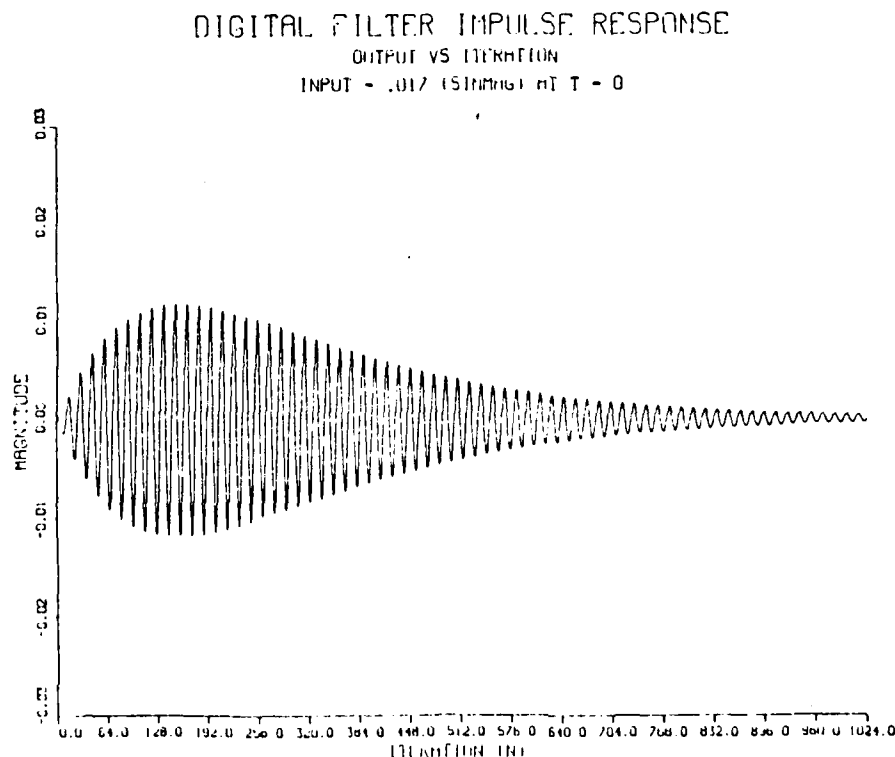


Figure 4.8 Digital Filter Impulse Response

In Figure 4.8 we observe the the response of the two stage difference equation filter for an input of 0.0172 applied at time $T = 0$. By counting the number of cycles which occur over a corresponding number of iterations, and realizing that the sample period is 0.11521 milliseconds we arrive at natural frequency which is very close to the expected 600 Hz. Additionally, it is apparently true that the response decays to zero with time, at least over the

Although we have ensured an input less than one volt this is not sufficient to guarantee that the post-processing value will not exceed the internal arithmetic limit. Internal arithmetic is limited to a range of values which cannot exceed -1.00000000 to +0.99999999. These 8 decimal place accuracies are established by the internal 25 binary bits (1 sign bit and 24 magnitude bits) available for arithmetic computations within the 2920. Actually the range of multiplicative inputs is only good to within 4 decimal places due to the scaling problem. But this will be seen to be more than adequate for our purposes.

To ensure that the processed values do not exceed one volt we scale down the digitally sampled input value by 64 by way of program step 44. The difference equation manipulations of the input value are then accomplished in program steps 47 through 130.

Digital arithmetic is performed in the 2920 by means of binary shifting and adding which is predicated on a transformation of coefficients to a nearest binary equivalent. The FORTRAN program and its output which performs this transformation is labelled CTRANS2 and is shown in Appendix I. Although a binary transformation does involve some approximation error, we see in the appendix that the worst case approximation of coefficients is still within .02 percent of the actual value. This is a relatively insignificant error.

D. A 2920 DIGITAL FILTER IMPLEMENTATION

Here we shall describe the particular 2920 software and hardware components which comprise the digital filter.

1. 2920 Assembly Language Program

In Appendix H we find the 2920 assembly language program which implements the two stage difference equation developed in Chapter 4. Recognizing the characteristics of the 2920 processor, it is instructive to review the programming devices which are brought to bear to realize this filter. We will proceed in the order in which these devices are used in the program. Appendix H should be consulted as reference for the discussion which follows. A detailed discussion of the 2920 Assembly Language should be consulted for particulars concerning the language [Ref. 11].

After initializing the DAR register we accept the input analog sample from the sensor microphone/preamplifier ensuring that the level does not exceed 1.0 volts. This limit is established by the voltage reference circuitry at pin #8 of the 2920. The input analog value is stored in the Sample/Hold register.

We then begin a sequence of steps, according to 2920 protocol, which accomplish the analog to digital transformation of the input value in the Sample/Hold register. This procedure is completed at program step 43 and the resulting digital value is then found in the DAR register.

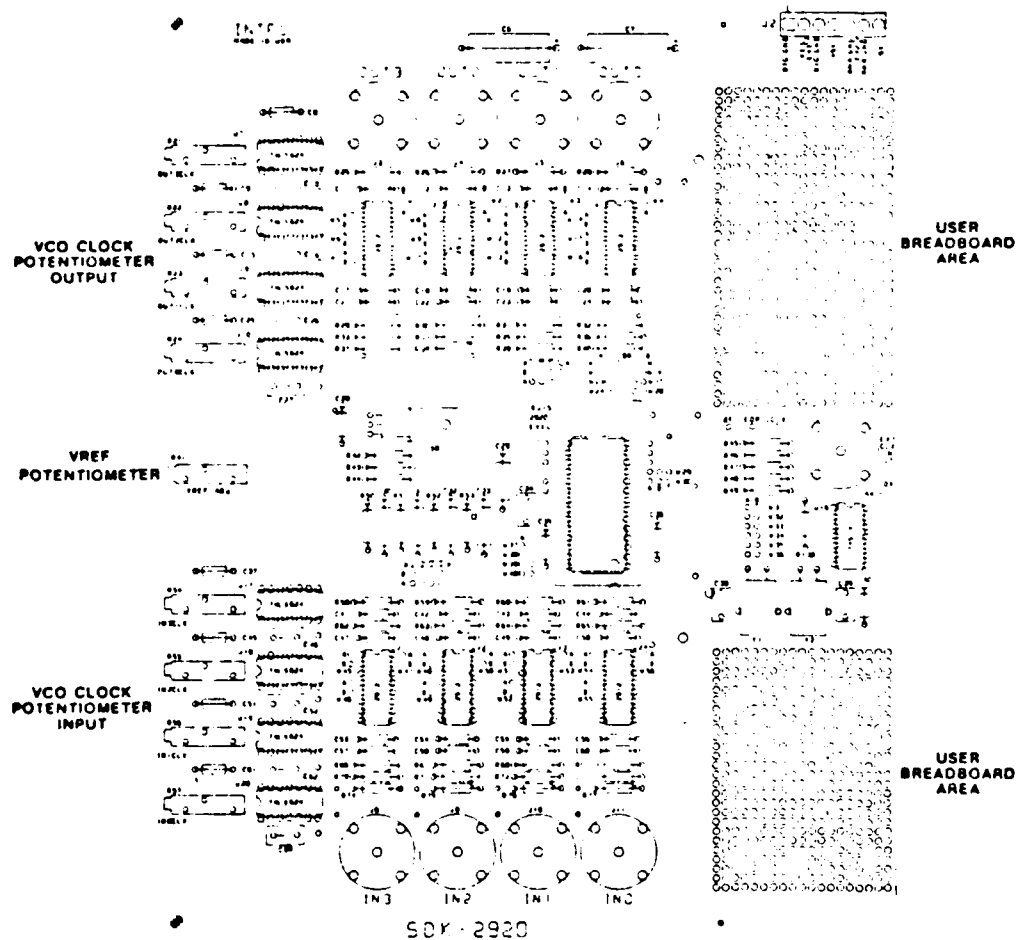


Figure 5.3 SDK-2920 Applications Board

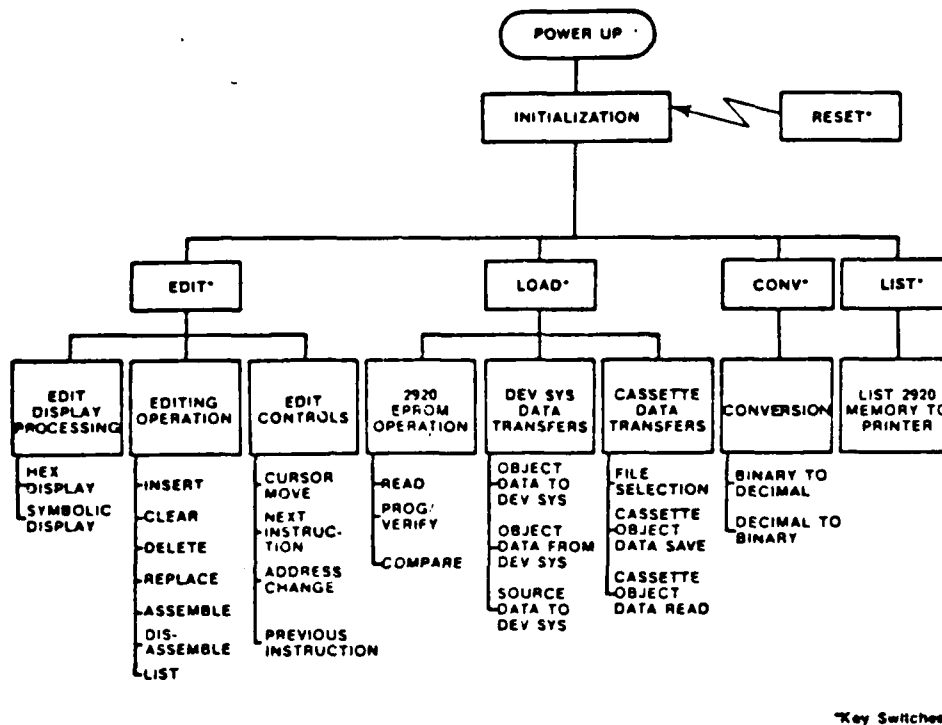


Figure 5.2 SDK-2920 Monitor Command Structure

of the SDK-2920 by examination of inputs and outputs. The applications board is shown in Figure 5.3 on the page following. Provisions are made on the board for assembly of either internal or external clocks, four input and four output channels with associated waveshaping circuitry, reference voltage development, and two user breadboard areas for specific applications development. Furthermore, TTL compatible output signals can be delivered to the output vice analog outputs if desired. We shall make use of this feature to send a signal detection pulse when the APU ignition is detected.

Applications software was developed using an Intellec Microcomputer Development System running a 2920 Assembly Program and Software Simulator. Transfer of 2920 software between the development system and the SDK-2920 is easily accomplished.

The SDK-2920 is physically divided into a development side and an applications side. The development side can be used to load, test and modify EPROM resident programs under 8085A microprocessor control. System control is accomplished with the use of a keypad monitor. The composition and hierarchy of the monitor command structure is shown in Figure 5.2 on the following page.

The applications side includes a prototype area for circuit construction and testing. It functions independently of the development side. After program development has taken place two methods may be used to accomplish program verification. The first method uses the Intel SM2920 Simulator Software to simulate the execution of programs written for the 2920. This simulator allows the use of symbolic references for changing and displaying all 2920 registers, flags and user-defined locations in program and memory storage. A trace feature also allows monitoring of selected parameters as they are changed under program control.

The second method of 2920 program verification is done by monitoring circuit performance on the applications side

these functions the analog section includes the following subsections:

- a four input multiplexer
- an input sample-and-hold circuit
- a D/A converter
- a comparator
- an output multiplexer with eight output sample-and-hold and buffer amplifiers.
- a special digital-to-analog (DAR) register which acts as an interface between the digital and analog sections.

C. THE SDK-2920 DEVELOPMENT SYSTEM

The SDK-2920 Development System is an integral component in the development of any applications package which uses at its core the INTEL 2920 Analog Signal Processor [Ref. 10]. Within the scope of the system are many development capabilities including

- Breadboarding: The breadboard is used to develop circuits for evaluation or prototype applications.
- Assembling and Editing: This feature is comprised of an assembler, disassembler, hexadecimal display, symbolic 2920 instruction display, and single keystroke entry of many 2920 instruction fields.
- 2920 EPROM Programming: The development board includes hardware and control elements necessary to program the 2920.
- Communications: The development also interfaces with Intel Developments Systems (such as the Inteltec Series) to pass object and source code listings of 2920 programs.

function, any applications program cannot make use of more than 192 instructions to process whatever number of input and output signals are being manipulated. But despite this restriction the power of the 2920 is evident. In our application we will only make use of a single input/output channel.

B. 2920 FUNCTIONAL DESCRIPTION

Figure 5.1 on the previous page details the block configuration of the 2920 [Ref. 9]. It is divided into the three major subsections described as follows.

The 192 x 24-bit Program Memory Section is a storage area implemented with EPROM. This section includes the instruction clock and timing circuits and program counter which control the operation of the entire device, including the other two sections.

The Arithmetic Section includes a 40 word by 25-bit scratchpad RAM and an arithmetic and logic unit (ALU). Both the RAM and the ALU are two port access devices. In the case of the ALU one of the ports is passed through a barrel shifter scaler. The function of the arithmetic section of the 2920 is to execute the commands dictated by the program memory.

The Analog Section performs A/D and D/A functions upon command from the program memory. In order to implement

In addition to the precision and speed of computation which the 2920 offers, it also allows for sequential processing of up to four separate input signals and eight analog output signals in a single program pass. This is of course dependent upon program complexity--regardless of

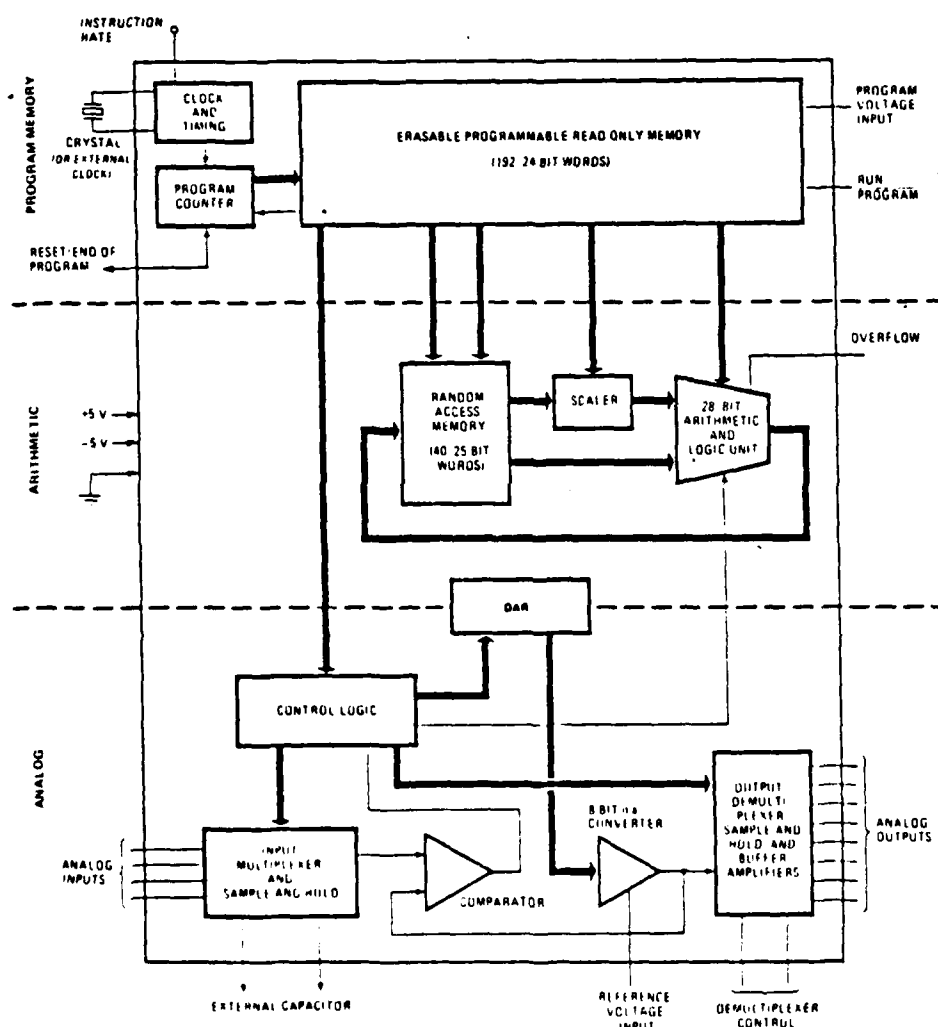


Figure 5.1 2920 Functional Block Diagram

Several logical conditions are allowed which affect program manipulation of data, but none will cause the processor to execute a program step out of sequence. In fact the only effective jump is performed at the last instruction which provides for a return to the beginning of the program loop. In this way the programmer may provide for an exact digital sample interval based upon program loop execution time. The shorter the program implementation loop the greater is the processor capacity to provide a faster sampling frequency.

The necessity of providing for an accurate sampling interval arises out of an understanding of the characteristics of the sampled analog signal being processed. Without an accurate clock interval, provided for in the 2920 by the program execution loop time, significant noise can be introduced in the system. Even small variations in the sampling interval can render the analysis useless through the introduction of intolerable measurement noise.

Each 2920 program instruction requires four clock cycles to execute. Given our nominal 6.666 MHz clock (the maximum allowed) the 2920 can therefore realize a maximum sampling frequency of 8.680 kHz over a 192 instruction program loop. This allows for a device bandwidth of greater than 4 kHz. Shorter programs naturally allow for a greater sampling frequency and thus higher device bandwidth.

V. THE INTEL 2920 ANALOG SIGNAL PROCESSOR

A. OVERVIEW

The INTEL 2920 Analog Signal Processor is actually a digital processor which is implemented to perform analog signal processing functions. Introduced in 1979, the 2920 system is centered about the 2920 single-chip microcomputer which is specially designed to process real-time analog signals. This single chip includes within its 28-pin DIP configuration sufficient hardware to provide 192 program memory locations, scratchpad memory, digital to analog (D/A) circuitry for up to four separate sampled inputs, analog to digital (A/D) capabilities for eight individual outputs, a digital pipeline processor capable of up to twenty-five bit accuracy, and input/output (I/O) control circuitry [Ref. 7: p. 3-1 through 3-2] . The 2920 is capable of implementing a wide variety of functions which rely upon sampled digital data techniques. We will use the 2920 to implement our matched filter design which will detect the APU start.

At the heart of the 2920's significant power is its on-board erasable programmable read-only memory (EPROM) which allows the user the convenience of customizing the 2920 for each intended application. Because the 2920 is a pipeline processor all program steps are performed sequentially without any conditions which may impact upon execution time.

filter. We shall see in the discussion which follows that the sampling frequency will be 8680 Hz. Thus our band of input frequencies is limited to less than one-half of this value, or 4340 Hz. Because our frequency of interest is 600 Hz we have considerable freedom in choosing the cut-off frequency of our anti-aliasing filter.

One option available to us is to design a low-pass filter with a rolloff which meets our needs. However, there are such filters commercially available which implement a compatible response which minimizes the effort required of the designer. One such filter is the INTEL 2912A which has been specifically included in the hardware kit we shall use to implement the digital filter we have just developed. This hardware implementation is the subject of Chapter 5.

output at 600 Hz is significantly less than at 590 Hz and even 575 Hz. This is indicative of both a steeper filter rolloff at frequencies greater than the center frequency and the effect of coefficient approximation which will be discussed more fully in the next chapter. The frequency response at both 500 Hz and 700 Hz is expectedly minimal but may not be usably low. If we find that the filter rolloff is not great enough and the response out of the passband is too great for our purposes then further design modifications may be undertaken. Accordingly, we could increase the order of our filter design. This would increase the number of filter stages in the analog implementation and therefore the complexity of that design. But, as we shall see, to a certain extent this additional filter complexity in the difference equation may be absorbed by the digital implementation we shall pursue without any increase in the hardware. These considerations will have to be examined more completely in the final analysis of the filter design effectiveness.

E. ANTI-ALIASING FILTER

When implementing a digital filter it is necessary to employ an analog input anti-aliasing filter to limit the band of input frequencies to less than half of the Nyquist sampling rate. This corresponds to the need to implement a low-pass filter at the input to the digital band-pass

data to recreate the frequency response. This is shown in Figure 4.9. The figure confirms a narrow band-pass filter function with a center frequency at approximately 585 Hz. This is very close to the design center frequency of 591 Hz and is, in fact, within the error of a single bar in this pattern representation.

2. Digital Filter Frequency Response

Now that we have confirmed the stability of our filter design we can proceed to examine the frequency response of the filter over the range of interest. In particular we shall examine the filter frequency response over several frequencies in the range of 500 Hz to 700 Hz. The FORTRAN programs which allow this examination are S22F and S22FG which are included as Appendix G to this thesis. Due to the number of output figures they will be left in the appendix and we shall only give a summary of their content.

The digital filter frequency response was examined for the following frequencies: 500, 575, 590.825, 600, 625 and 700 Hz. The 590.825 simulation was chosen because this is the design center frequency (due to pre-warping) and we wish to confirm an output maximum amplitude at this frequency. From the figures in the appendix it is easy to see that the filter does in fact yield the response we desire. The maximum output amplitude does occur for the expected frequency, although the output at 575 Hz does not diminish appreciably from this value. However, we observe

approximately one-tenth of a second represented by the duration of the overall sample period in the figure. To confirm this suspicion we can carry out the impulse response for a substantially longer period of time, say over one full second, or approximately 8192 iterations. The results of this computation are shown in the output of S22I in the appendix. They confirm the occurrence of the maximum amplitude of impulse response output at the xxxth iteration which is what we observe in the figure.

Having realized the digital filter impulse response output we can perform a discrete Fourier transform of this

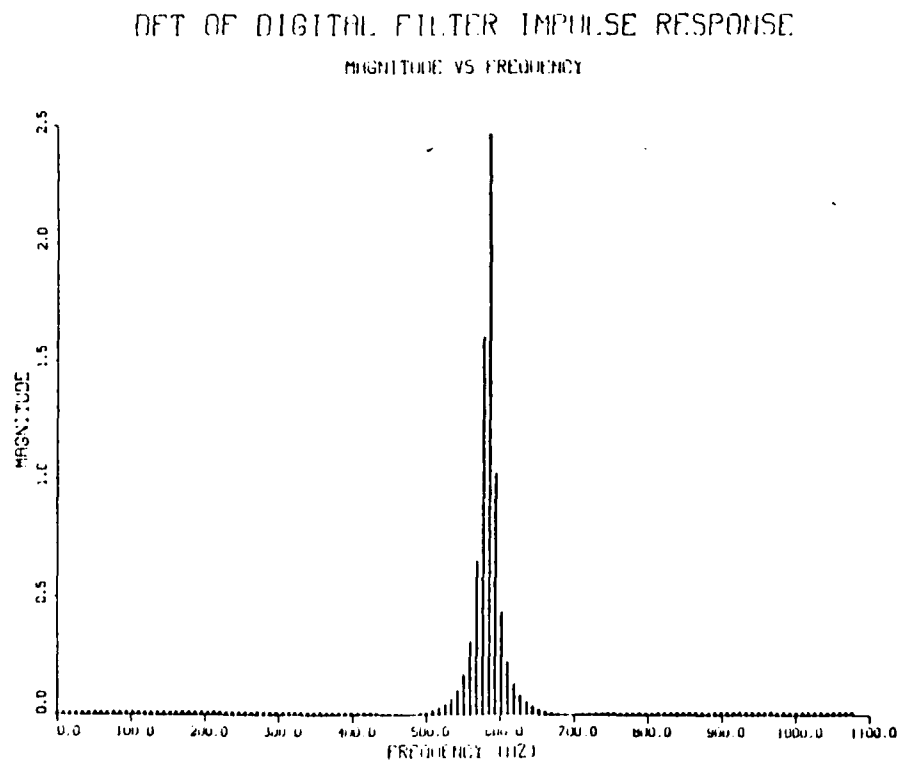


Figure 4.9 Discrete Fourier Transform of
the Digital Filter Impulse Response

After the difference equation implementation in each program pass we are left with a binary value in the DAR register which corresponds to the program output for that pass. We have the option of providing a certain amount of linear output gain by an appropriate shift of the output binary value now in the DAR. In program step 132 we provide a gain of four by a left shift of two binary positions. We output this value to channel 0 in steps 139 through 142.

The final program manipulation occurs in program steps 143 to 150. Here we perform a serial register shift of present program pass values in preparation for the next program pass. Program step 191 is the final executable statement which returns us to step 0 for the next pass. The entirety of the 2920 operation consists of an endless loop of these instructions.

2. 2920 Hardware Implementation

The 2920 contains an EPROM which is loaded with the hexadecimal code which is equivalent to the assembly language program just described. However, there are several other component devices which are integral to the operation of the 2920. The relation of these devices to the 2920 will now be described. A graphical schematic of these components appears in Figure 5.4.

At the input side of the 2920 an anti-aliasing filter is realized by using a 2912A which actually contains two filters which are cascaded together. This configuration

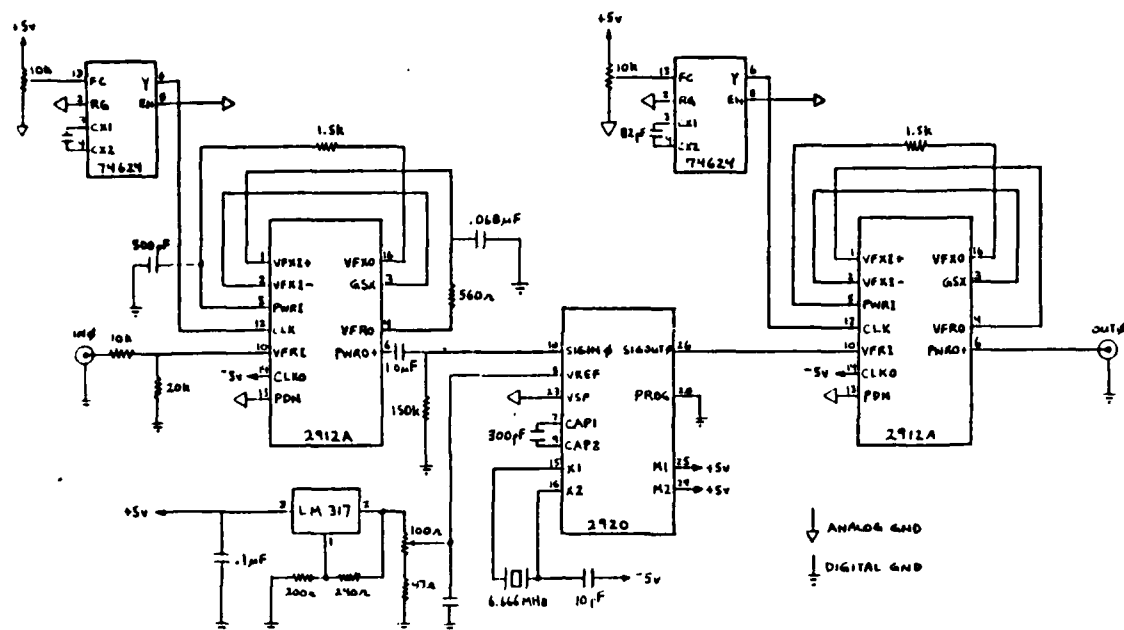


Figure 5.4 2920 Digital Filter Schematic

provides 54 dB of input dynamic range and a nearly flat response for frequencies less than 3 kHz. There a steep roll-off commences and at about 4 kHz the cascaded filter combination provides over 30 dB of attenuation. This supports the Nyquist frequency limit which is 4.34 kHz in this application.

The 2912A is a pulse code modulated filter which requires a clocking input to realize its filter function. This is provided by the 74624 at its input.

At the output of the 2920 another 2912A is employed in identical configuration and now provides a reconstruction filter for our application. This filter smooths the output

of the 2920 to provide an analog signal for follow-on logic discrimination as shown earlier in Chapter 3.

A 2920 option which is not demonstrated here yet will be employed in final filter configuration is to obviate the need for external signal conditioning by allowing program discrimination of the output value and thus providing a processed TTL signal output. This eliminates the need for the external circuitry shown in Figure 3.8 and therefore represents one significant advantage of the 2920 digital design over the analog implementation.

E. 2920 DIGITAL FILTER IMPLEMENTATION RESULTS

We will now proceed to demonstrate the results of the 2920 digital filter implementation in much the same manner as the presentation which accompanied the analog filter design in Chapter 3. We begin with a photograph of the digital filter frequency response to a ramped sinusoidal input. This is shown in Figure 5.5. The same method was used to generate the sweep oscillation although the range of sweep is not identical to that employed in generating Figure 3.9. The result is that we cannot guarantee the narrow bandwidth of this digital filter relative to its analog counterpart by this means alone. The intent is, as before, only to demonstrate that a narrow band-pass filter response is generated.

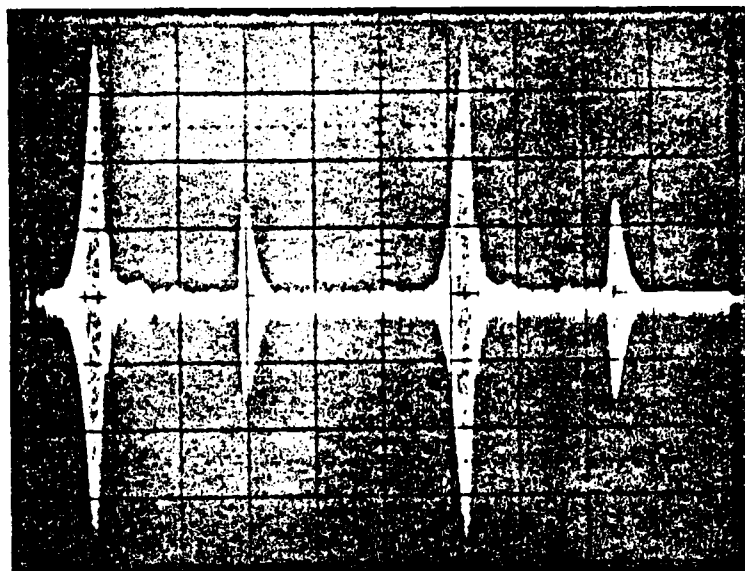
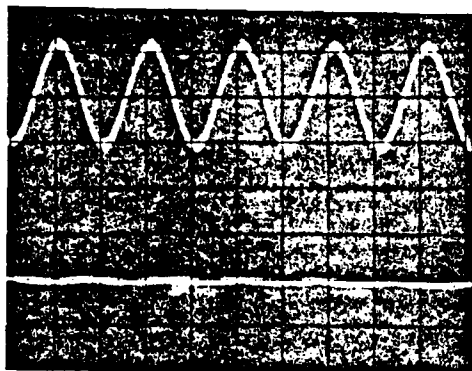
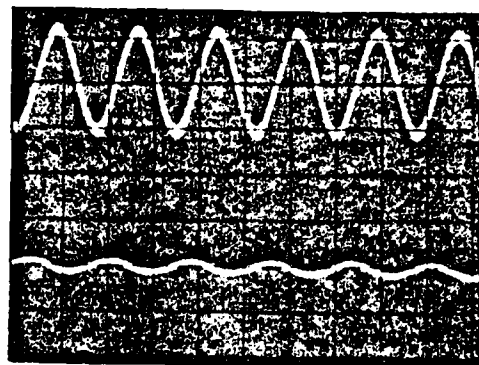


Figure 5.5 Digital Elliptic BPF Frequency Response
(Photograph of Actual Filter Amplitude Response
to a Ramped Sinusoidal Input)

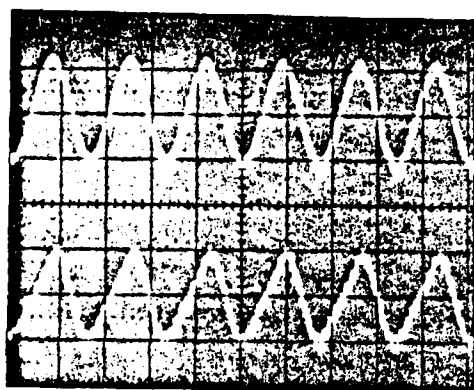
To confirm the operation at the desired band-pass center frequency we next apply discrete sinusoidal inputs to the digital filter at various frequencies arrayed about 600 Hz. The result is the digital analog to Figure 3.10 which is shown here as Figure 5.6. The scale is maintained as in Figure 3.10. The input frequencies are at about double the amplitude of the analog filter to ensure proper operation. This implies that despite the relative immunity of the digital filter to input amplitude variations we must nonetheless provide an input above approximately 100 millivolts peak-to-peak. However, once above this threshold



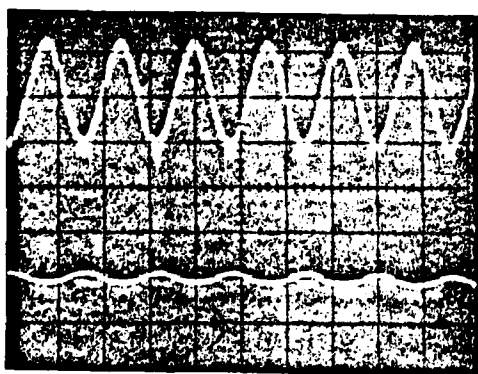
a) 500 Hz



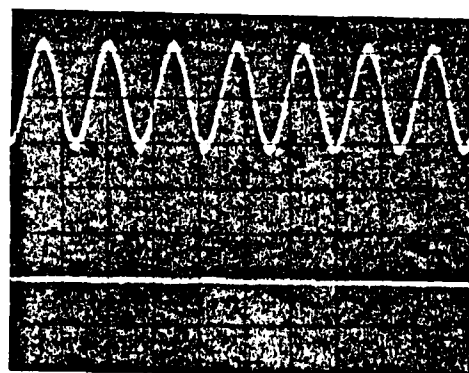
b) 575 Hz



c) 600 Hz



a) 625 Hz



b) 700 Hz

Figure 5.6 Digital Elliptic BPF Frequency Response
 Upper trace (Input): 50 mV/div scale
 Lower trace (Output): 1.0 V/div scale

value the digital filter provided a relatively undistorted and largely constant amplitude output up to an input amplitude of over 5 volts peak-to-peak (and this despite the 1 volt reference level of the input). Figure 5.6 thus confirms the center frequency maximum at a value near 600 Hz and a steep roll-off on either side of this value.

VI. ALTERNATIVE FILTER CONCEPTS

The preceding development was based upon techniques used in implementing an Infinite Impulse Response (IIR) digital filter. Simply stated, an IIR filter realizes its output based upon the values of all present and previous inputs and outputs. In other words, feedback is employed in an IIR design.

In the general case, an IIR filter will have M finite zeroes and N finite poles. The zeroes of $H(z)$ can be anywhere in the z -plane but the poles must lie within the unit circle to guarantee stability. In the case we have developed, a digital filter realization derived from an analog design, the order of M must be less than or equal to N . This describes an N th order digital filter.

The hardware implementation of an IIR design usually involves the cascading of elemental single pole filters with double complex pole filters. These elements are derived from the original transfer function using a partial fraction expansion separation scheme.

There are other methods for realizing the filter we desire other than the a priori scheme we have developed so far. These generally use the input signal itself as a basis for the filter transfer function coefficients and involve an

adaptive evaluation of the proper coefficients which yield the desired filter response.

A. A WIENER FILTER DESIGN--THE ADAPTIVE LINEAR COMBINER

The Adaptive Linear Combiner (ALC), shown in Figure 6.1, forms the basis for the Adaptive Filter design we shall now discuss [Ref. 12]. An input analog signal may be digitally sampled in accordance with the Nyquist criterion and we may then apply N sequential elements of that sample block to

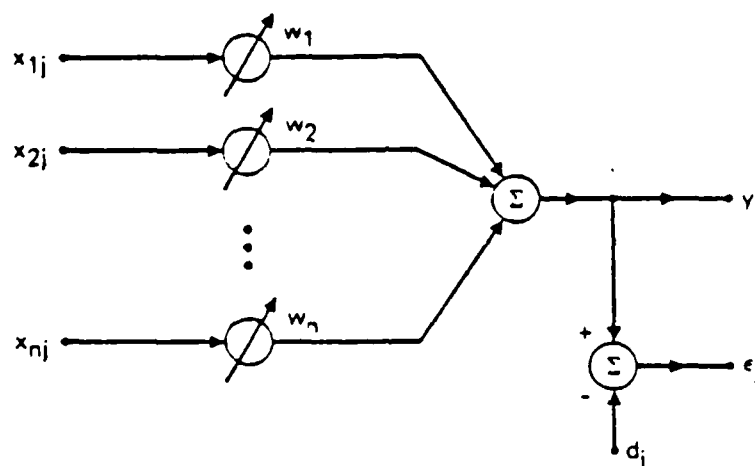


Figure 6.1 The Adaptive Linear Combiner

the ALC inputs. These inputs can be easily derived from a tapped delay line which cascades sample values along in sequential storage for processing. This scheme lends itself well to implementation in a processor such as the 2920 which is designed to accept sequential values by way of its component A/D converter and RAM storage.

The set of measurements x_{nj} is multiplied by a corresponding weighting term W_j , and the results then summed to yield the output y_j . This output is then compared to a desired signal value for that instant and the difference between them constitutes an error signal ϵ_j . The objective of the ALC is to determine W_j so as to minimize ϵ_j for each set of sampled inputs and thus realize the weighted sum of input signals that best matches the desired response.

1. Theoretical Foundations

At the n th instant of time the output of the Non-Recursive Wiener ALC, $y(n)$, is given by [Ref. 13]:

$$\begin{aligned} y(n) &= \sum_{j=0}^N x(n-j)W_j \\ &= W_0x(n) + W_1x(n-1) + \dots + W_Nx(n-N) \end{aligned}$$

which may be written in matrix form as

$$= \underline{W}^T \underline{X}$$

or equivalently

$$= \underline{X}^T \underline{W}$$

where T represents the matrix transpose operator, the set of $N+1$ weights is denoted by

$$\underline{W}^T = [W_0, W_1, \dots, W_N]$$

and the set of present and N previous inputs is given by

$$\underline{X}^T = [x(n) \ x(n-1) \ \dots \ x(n-N)]$$

The error signal $e(n)$ for time n is given by

$$e(n) = d(n) - y(n)$$

$$= d(n) - \underline{W}^T \underline{X}$$

and the square of the error (using the latter matrix notation) by

$$\begin{aligned} e^2(n) &= e(n) \cdot e^T(n) \\ &= [d(n) - \underline{W}^T \underline{X}][d(n) - \underline{X}^T \underline{W}] \\ &= d^2(n) - 2d(n)\underline{X}^T \underline{W} + \underline{W}^T \underline{X} \underline{X}^T \underline{W} \end{aligned}$$

The mean square error, obtained by taking the expected value of this last equation, is given by [Ref. 13]

$$E[e^2(n)] = E[d^2(n)] - 2E[d(n)\underline{X}^T] \underline{W} + \underline{W}^T E[\underline{X} \underline{X}^T] \underline{W}$$

Defining the vector Φ_{xd} as the cross-correlation between $d(n)$ and \underline{X} then yields

$$\Phi_{xd} = E[d(n)\underline{X}]$$

$$= E[d(n)x(n), d(n)x(n-1), \dots, d(n)x(n-N)]^T$$

The input auto-correlation matrix Φ_{xx} is defined as

$$\Phi_{xx} = E[\underline{XX}^T]$$

which may be written in expanded notation as

$$= \begin{bmatrix} x(n) \\ x(n-1) \\ x(n-2) \\ \vdots \\ x(n-N) \end{bmatrix} [x(n) \ x(n-1) \ \dots \ x(n-N)]$$

Now if we carry out the indicated vector multiplication we arrive at the following result [Ref. 13]

$$= \begin{bmatrix} x(n)x(n) & x(n)x(n-1) & \dots \\ x(n-1)x(n) & x(n-1)x(n-1) & \dots \\ \vdots & \vdots & \\ \vdots & \vdots & \\ & & x(n-1)x(n-1) \end{bmatrix}$$

And thus we arrive at the following form of the input correlation matrix

$$= \begin{bmatrix} \phi_{xx}(0) & \phi_{xx}(-1) & \dots & \phi_{xx}(-N) \\ \phi_{xx}(1) & \phi_{xx}(0) & \dots & \phi_{xx}(1-N) \\ \vdots & \vdots & \ddots & \vdots \\ \phi_{xx}(N-1) & \phi_{xx}(N-2) & \dots & \phi_{xx}(-1) \\ \phi_{xx}(N) & \phi_{xx}(N-1) & \dots & \phi_{xx}(0) \end{bmatrix}$$

In order to find the optimal weight vector, \underline{W}^* , we can differentiate the mean square error function with respect to the weight vector \underline{W} to yield

$$\frac{d(e^2(n))}{d\underline{W}} = -2[\underline{\Phi}_{xd} - \underline{\Phi}_{xx}\underline{W}]$$

The optimal weight vector, \underline{W}^* , generally called the Wiener weight vector, is obtained by setting the quantity in brackets equal to zero. This results in

$$\underline{W}^* = \underline{\Phi}_{xx}^{-1} \underline{\Phi}_{xd}$$

The objective of processes involving the ALC is to find a solution to this equation. In fact we may employ an adaptive algorithm which uses the error signal, $\epsilon(n)$, (generated for each instance of filter inputs), as the basis

for modifying the filter weights until a minimum error is attained for a particular input block. This describes the Adaptive Transversal Filter shown in Figure 6.2 [Ref. 12].

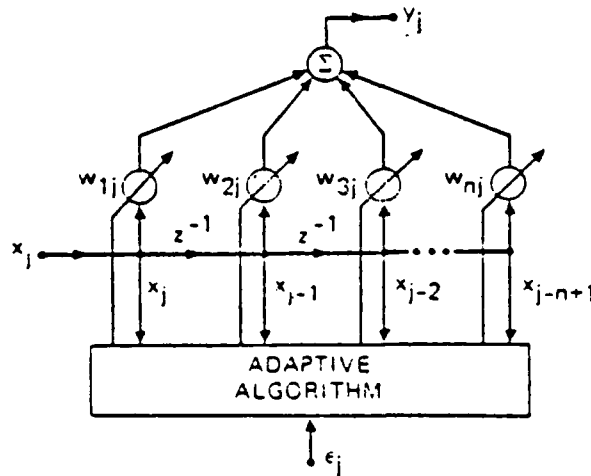


Figure 6.2 The Adaptive Transversal Filter

The Adaptive Transversal Filter (ATF) is a Finite Impulse Response (FIR) filter owing to the lack of direct feedback from output to input. If we employ a tapped delay line at the input to the ALC which comprises the ATF the form of the input vector becomes a finite number of delayed elements of the input signal. It is therefore easy to see that the impulse response of the ATF is just the sequence corresponding to the elements of the weight vector, \underline{W} . Such a filter can have any impulse response of length less than or equal to its own length. Allowing for an ideal unlimited length we could realize any impulse response at

all, and thus any frequency response. Practically, however, we are limited by filter complexity, error due to misadjustment, and an adaptive time constant which corresponds to filter length.

Thus we have a means of generating the desired filter response by applying the very signal we wish to detect. If we apply a digital series of samples taken from an analog reference signal we can realize the filter weights which will provide our desired signal output stream at a later time.

Thus the idea is to sample the analog recording of the APU noise in the cargo bay prior to launch and to apply that input series of data elements to an ATF to realize the filter weights. We may then build a 2920 circuit which uses these weights as filter coefficients to provide our filter response.

2. A Software Simulation

As an example of this methodology we will now present an elementary simulation which was performed for an input which consisted of an equal amplitude application of the three fundamental frequencies of interest: 600 Hz, 1200 Hz and 1800 Hz. We chose to simulate an Adaptive Transversal Filter of fourth order which therefore consists of four weights.

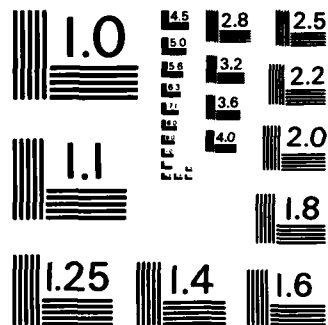
One example of a software implementation which is designed to arrive at the four desired filter weights is

A MATCHED FILTER ALGORITHM FOR ACOUSTIC SIGNAL
DETECTION(U) NAVAL POSTGRADUATE SCHOOL MONTEREY CA
D W JORDAN JUN 85

UNCLASSIFIED

F/G 17/1

NL



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS-1963-A

shown in Appendix J as FORTRAN program FIR4. In this program we begin with trial weights and a range of upper and lower bounds. By repeated application of a library coefficient optimization routine (BOXPLX--also included in the appendix) we arrive at a set of four optimal weights within the bounds specified.

The result of this simulation is revealed through application of the FORTRAN program FIR4SIM which is included as Appendix K to this thesis. This result is shown in Figure 6.3. The input analog signal (indicated by the solid line) is a portion of the combined signal corresponding to

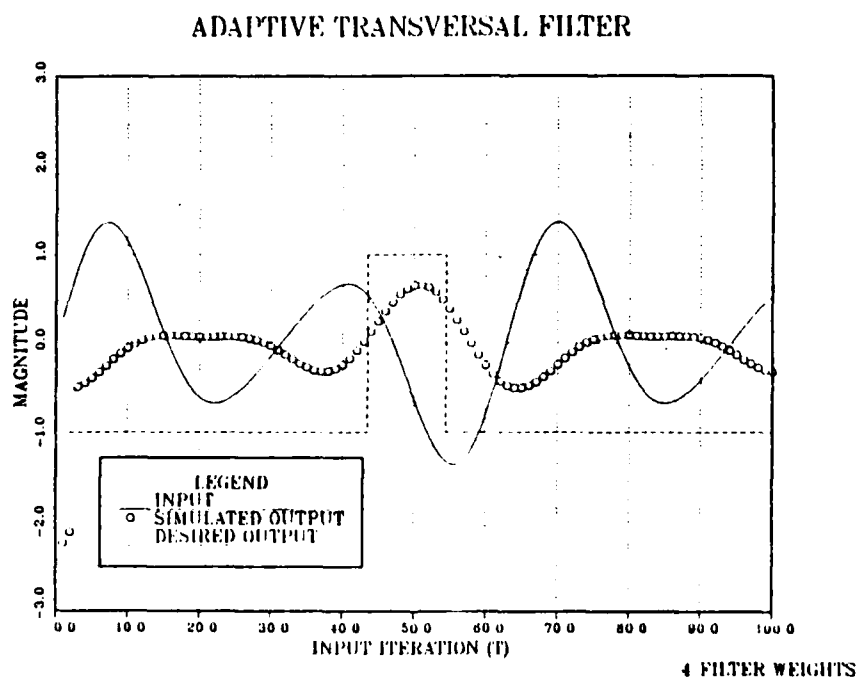


Figure 6.3 An Adaptive Transversal Filter Simulation

the three fundamental frequencies mentioned previously. The desired output (shown by the dashed line) is chosen to be a continuous -1 unless the signal of interest is detected. In that case the desired output jumps to +1.

As indicated in Figure 6.2 the Adaptive Algorithm samples the input signal and uses the successive present and three previous samples to arrive at the desired filter weights which will accomplish the task of signal discrimination. In the algorithm implemented by FIR4 of Appendix J we arrived at the following filter weights.

$$\begin{aligned}w_1 &= -7.5060358 \\w_2 &= 7.5403662 \\w_3 &= 4.7097464 \\w_4 &= -5.3987589\end{aligned}$$

In Figure 6.3 we see that over 100 sample output iterations these weights resulted in an output which was at or near zero or below with a significant rise above 0.5 near the desired region. This approximates the filter response which would allow a useful discrimination of the desired signal by detection above a threshold floor (say 0.5 in this example).

This is by no means intended to be an exhaustive discussion of this approach to a matched filter design, but merely a consideration of an alternative approach which might be taken to realize a useful filter.

B. AN ANALOG SPEECH PROCESSING SCHEME

A further alternative which may be considered involves the use of commercially available speech processing microcircuits which often use Linear Predictive Coding schemes as the basis for their discriminant filters.

The current state of the art in speech recognition technology does not permit even the most sophisticated (and large) devices to recognize but several hundred words of vocabulary. The breakthroughs are most often in the arena of overcoming the speaker dependent nature of the simpler systems. However, all systems, be they single chip processors or multi-cabinet devices, do have the capability to analyze an audio input signal (conventionally this is speech of course) and to characterize the nature of changes in the formant composition over time.

Because the signal we wish to identify is in the audio spectrum it seems a logical idea to consider that a speech recognition device may prove usable for our purposes. In fact, Interstate Electronics Corporation now markets a single chip voice recognition device (VRC008) which is capable of reliable and independent recognition of up to eight words or phrases which are stored in its vocabulary. While this may seem a minimal vocabulary it is a remarkable ability for a single chip device.

The Interstate VRC008 is capable of being trained to recognize words or phrases of up to 1.2 seconds duration.

To implement an audio signal recognition scheme would require that we somehow "sample" our input audio environment in discrete blocks of about one second apiece. Thus we would simulate a discrete utterance which could be processed by the circuit. In the absence of the characteristic APU signal the device would register no recognition of the input sample. But after APU ignition it is reasonable to assume that the device would treat the APU signal as a recognizable "word" it had previously been taught.

While this approach may seem at first to be a promising one we must also consider the drawbacks, especially in view of the technically simpler approaches we have reviewed so far. First is the cost. Although the Interstate chip is by itself a relatively inexpensive device (on the order of ten dollars in quantity), by itself it is also useless without a twenty-five thousand dollar training and development tool. The intent of the manufacturer is that the cost of the development tool will be amortized by the consumer over a sizable run of usable end devices. Our application does not lend itself to mass production and the cost therefore becomes prohibitive. This is especially true in view of the cost of the simpler technologies discussed in previous chapters which have given us useful designs.

Another drawback of a speech recognition approach is the level of technical complexity versus guarantee of successful results. Unless we use a device of modular circuit size or

smaller we risk an excess of power and space consumption. But the state of the art in speech recognition is such that accuracy of recognition is roughly related to the size of the device (although it is directly the vocabulary size which is the truly overwhelming factor here). The Interstate VRC008 claims only an 85 percent accuracy of recognition which is low by the standards of other speech recognition devices. This is the price one pays for small size.

The important point is that our signal of interest is characterized by extremely well-defined and stationary spectral components. This fact allows the use of cheaper and more traditional methods of signal processing and filter design. Were our signal of a rapidly time varying nature then a purely analog approach would be impossible, and even a digital approach would prove difficult if not infeasible. It is then that methods of linear predictive coding which form the basis of speech recognition would become one of a very few viable alternatives.

VII. CONCLUSION

In this thesis I have considered several approaches to the problem of designing a matched filter for the detection of the acoustic signal which characterizes the Shuttle Auxiliary Power Unit. The Analog and Digital IIR filter approaches were treated in some detail, while the Weiner FIR and Voice Recognition methods were given less attention. My purpose was not to present an exhaustive treatise on the subject of filter design, but rather to describe various ways in which a particular problem might be approached.

It is not coincidental that the order of presentation of the considered methods should conform to the chronological introduction of these sciences to the engineering community. As may be expected, the facility with which each of these methods is employed is proportional to their general familiarity among engineers. The analog approach considered first is the best established method of filter design. Not surprisingly, this method is supported by a wealth of literature. Despite this ample documentation, at best the analog approach to filter design is an inexact science which is largely dependent upon the degree to which one is able to characterize the signals we wish to manipulate. Often, however, we have excellent knowledge of these signals, and thus the analog approach to filter design remains a

completely reasonable and certainly cost effective approach to simple filter designs.

The APU signal of concern to this study was such a signal. Its signature was stationary over time and could be reliably found at amplitudes well above the noise threshold. The dominant component at 600 Hz was of quality sufficient to preclude examination of sub-dominant spectral harmonics at higher frequencies. The fact that a well-defined signal was evident allowed for a design which emphasized the simplicity of the analog approach.

Mention should be made of the obstacles which did impede the final analog design. Because an analog filter serves only to attenuate those signal component spectral elements out of the passband, but does not eliminate them, it is necessary to know the range of amplitude which may be expected of the sensor microphone output. For a given amplitude of signal input which varies little within the range of input frequencies it may be reliably expected that the analog bandpass filter would reject the frequency components outside of the narrow passband. But if the spectral components were grossly disparate in their amplitude and a component out of the passband were received which was significantly above the amplitude expected of the 600 Hz center frequency, then it is possible that the component out of the passband would be passed regardless of

the filter attenuation. This demonstrates the need we have to know the nature of the input signal.

One approach to this problem is to increase the attenuation of the filter. But this does not guarantee signal component rejection out of the passband. The solution for an analog approach lies in Automatic Gain Control (AGC) at the sensor microphone input to the filter. In this manner we can ensure a dynamic range of input signal which is within the limits of filter discrimination. This implies a careful selection of a microphone and preamplifier combination which in turn implies a similarly careful understanding of the dynamic range of the input signal. Indeed, these considerations continue to be the most vexing aspects of a useful final design. The actual dynamic range of the signals recorded on tape was unreliable due to the number of intermediate and indeterminate dubs which the tape underwent prior to our acquisition of a copy.

Furthermore, and even more importantly, the ultimate placement of the sensor microphone in the Shuttle cargo bay will have considerable effect upon the nature of the signal available for discrimination. It will also tell significantly on the dynamic range. This factor will impact upon any chosen filter design regardless of the algorithm selected. Thus in the analog case we must design for a wide dynamic range and provide AGC which yields a narrower range of amplitude input into the filter.

Much of this problem is overcome with the digital filter implementation developed in Chapters 4 and 5. At the foundation of the digital design is a frequency domain scheme whose output is less dependent upon input amplitude variations than the frequency components of the input signal. In fact using an EPROM based filter design such as afforded by the INTEL 2920 we enjoy considerable flexibility in tailoring the range of allowable inputs and outputs through careful selection of program parameters. The limitations are rather imposed by the noise level at the low end and the power limit at the high end.

There are several drawbacks to the digital filter which bear mentioning. The foremost drawback is cost relative to the analog filter. The design presented in Chapter 5 was dependent upon the SDK-2920 Development Kit which is a thousand dollar item. This is the minimum hardware which is necessary to develop a 2920 signal processing design. However, to support any sort of a sophisticated development requires the INTEL Inteltec Development System with associated software. This quickly elevates the expense of the system to a range of tens of thousands. Of course there are certainly more uses for the Inteltec system than simply a 2920 development application, so this expense can be amortized over those additional uses. But the 2920 applications software which supports the Inteltec system is a four thousand dollar expense by itself.

This fact proved significant to the digital design when the simulator software was found to have a bug in it. When the original disks could not be located it was then deemed more practical to develop an application specific simulation on a mainframe computer instead of purchasing a replacement package from INTEL. This meant an additional expenditure of time of course, and was only successful in showing the adequacy of the specific 2920 filter implementation algorithm. However, without the 2920 Simulator software package effective troubleshooting was made significantly more difficult. Nonetheless, as indicated in the results of Chapter 5, a successful 2920 implementation was accomplished without a fully healthy simulator. With it the design process would have been considerably more efficient.

An additional consideration is the complexity of the digital design over the analog approach. This is due in large part to the availability of resources which support an analog design relative to the novel approach represented by a state-of-the-art signal processing chip. However, the complexity of a signal processor application is often far outweighed by the considerable flexibility which it provides. One must not forget the power of the 2920 (witness its ability to incorporate all of the hardware elements of the follow-on logic circuitry required in the analog design in but a dozen or so lines of 2920 assembly language code) and weigh this against the short term

inconvenience of having to become acquainted with a new approach. Once mastered the significant ability of a signal processing device make an analog approach to any complex filter design seem archaic. In addition the fewer actual circuit components required in an EPROM based device means significant savings in power consumption. This is an especially noteworthy item when considering an electronic device for a space application.

For the purposes of this thesis I must admit that the 2920 was certainly fun to work with. The literature is sketchy in spots and several calls to technical support at INTEL were needed to resolve some issues and errors. But overall the 2920 certainly provides the researcher with a significant amount of flexible and powerful signal processing ability.

The significant advantage of implementing a digital filter over the analog design is the relative immunity to the variations in input amplitude. This was a crucial consideration in the development described in Chapter 5 and by itself would account for the choice of the digital design over the analog approach. When coupled with the further advantages of lower power consumption, less physical space required and considerable flexibility in accommodating future changes without the need for hardware modification, the digital approach implemented in a powerful signal processor becomes an irresistible filter design option.

In Chapter 6 we considered two other approaches to the APU signal detection problem. Unfortunately a lack of time prohibited a serious examination of these additional approaches. Both are well-founded and represent the leading edge of signal processing technology. Given a requirement for detection of a more complex signal than we considered in this paper, these latter methodological options could well represent the only viable means of processing a time-varying signal in the acoustic spectrum.

APPENDIX B

POWER SPECTRAL DENSITY PLOTS OF THE SHUTTLE CARGO BAY PRE-LAUNCH ACOUSTIC ENVIRONMENT

Legend for the Graphical Output on the Following Pages

Shuttle Flight Number: STS-2, STS-3 or STS-4

Microphone Identification: 9405, 9219 or 9403

Sampled Interval Relative to APU Power-Up: PRE or POST

All PSD Sources are from the Original Aerospace Tape
Copies (labelled ORIG)

Narrow Band Analysis (N=40 Samples)

Hanning Weighting

5.0 Volt RMS Front End Limiter

Gain: 10 dB per division

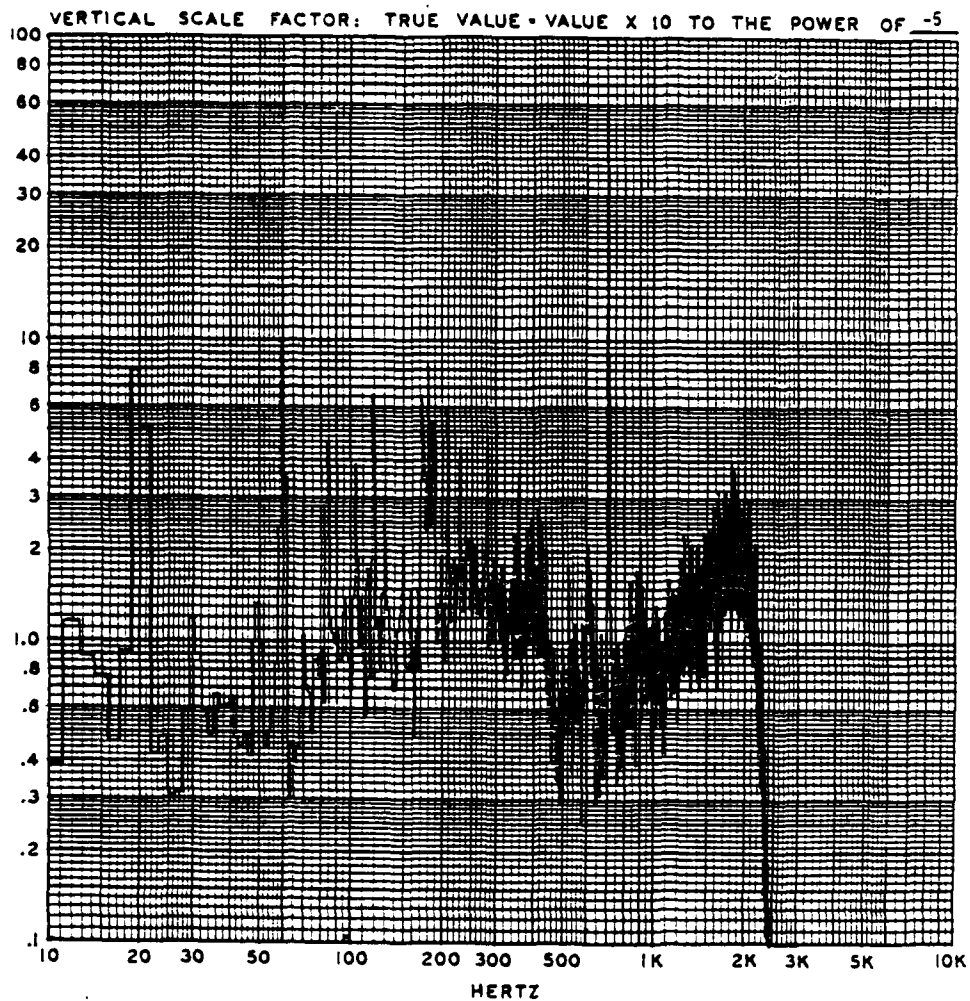
Cursor Point Label: X (Hz) and Y(B) (Engineering Units)

Scale: Linear Ordinate (0-2000 kHz)
Logarithmic Abcissa (10^1 to 10^4 EU)

RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

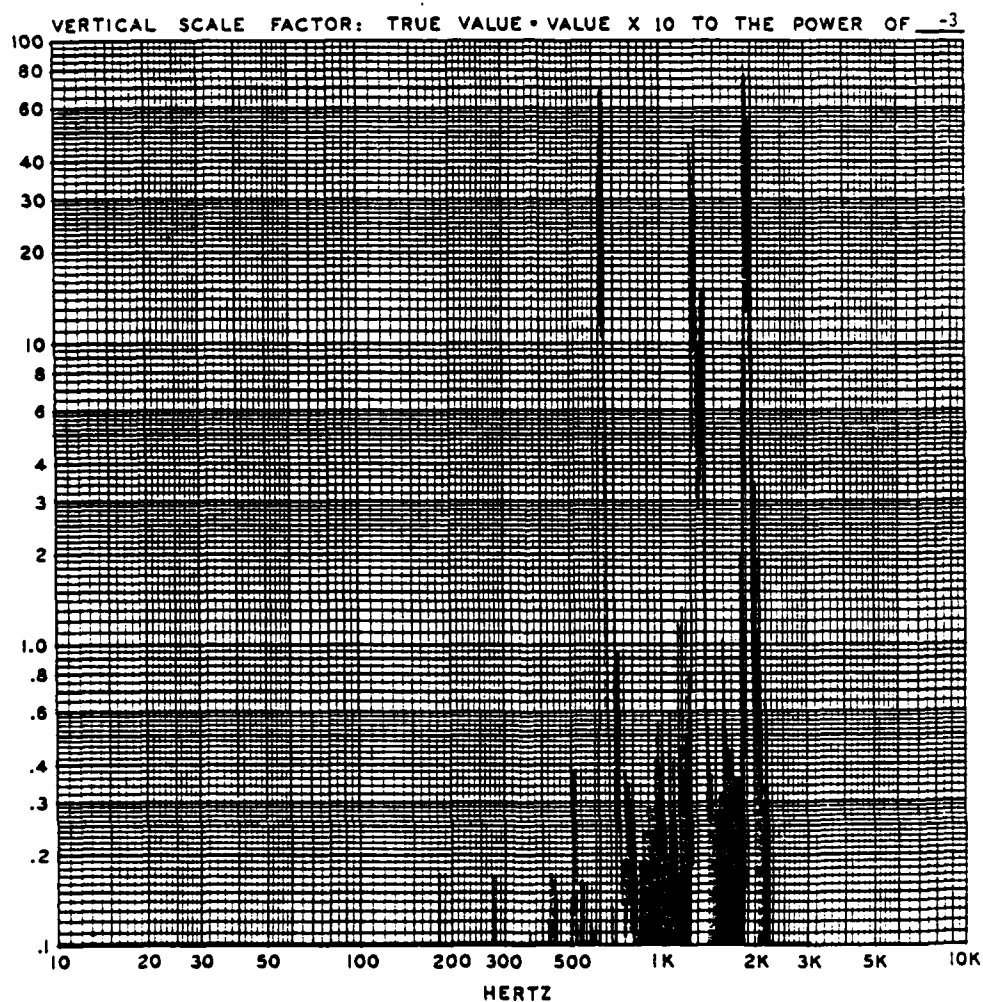
PARAMETER 00281A Noise Floor
OVERALL RMS VALUE 0.1905 G's
ENGINEERING UNITS 30.0000 G's
START TIME: (HR: MIN: SEC) 258/12:04:20
SAMPLE RATE (S/SEC) 6134.9687

TEST DATE 15 September 81
ANALYSIS BW 2.2467 Hz
ANALOG L.P. FILTER BW 2300.0000 Hz
DEGREES OF FREEDOM 9.0000
PROCESS DATE 18 September 81



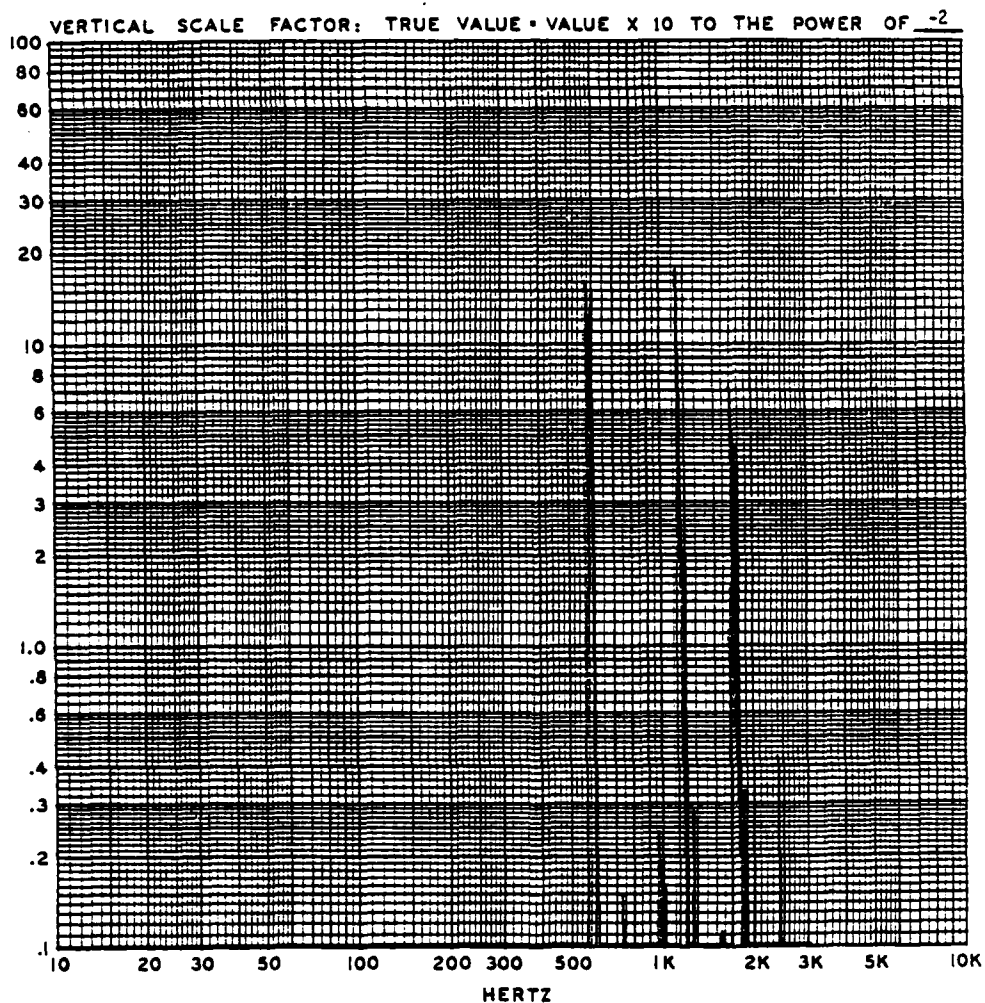
RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

PARAMETER <u>00281A</u>	TEST DATE <u>15 September 81</u>
OVERALL RMS VALUE <u>2.7251 G's</u>	ANALYSIS B W <u>2.2467 Hz</u>
ENGINEERING UNITS <u>30.0000 G's</u>	ANALOG L.P. FILTER B W <u>2300.0000 Hz</u>
START TIME: (HR: MIN: SEC) <u>258/12:03:20</u>	DEGREES OF FREEDOM <u>9.0000</u>
SAMPLE RATE (S/SEC) <u>6134.9687</u>	PROCESS DATE <u>18 September 81</u>



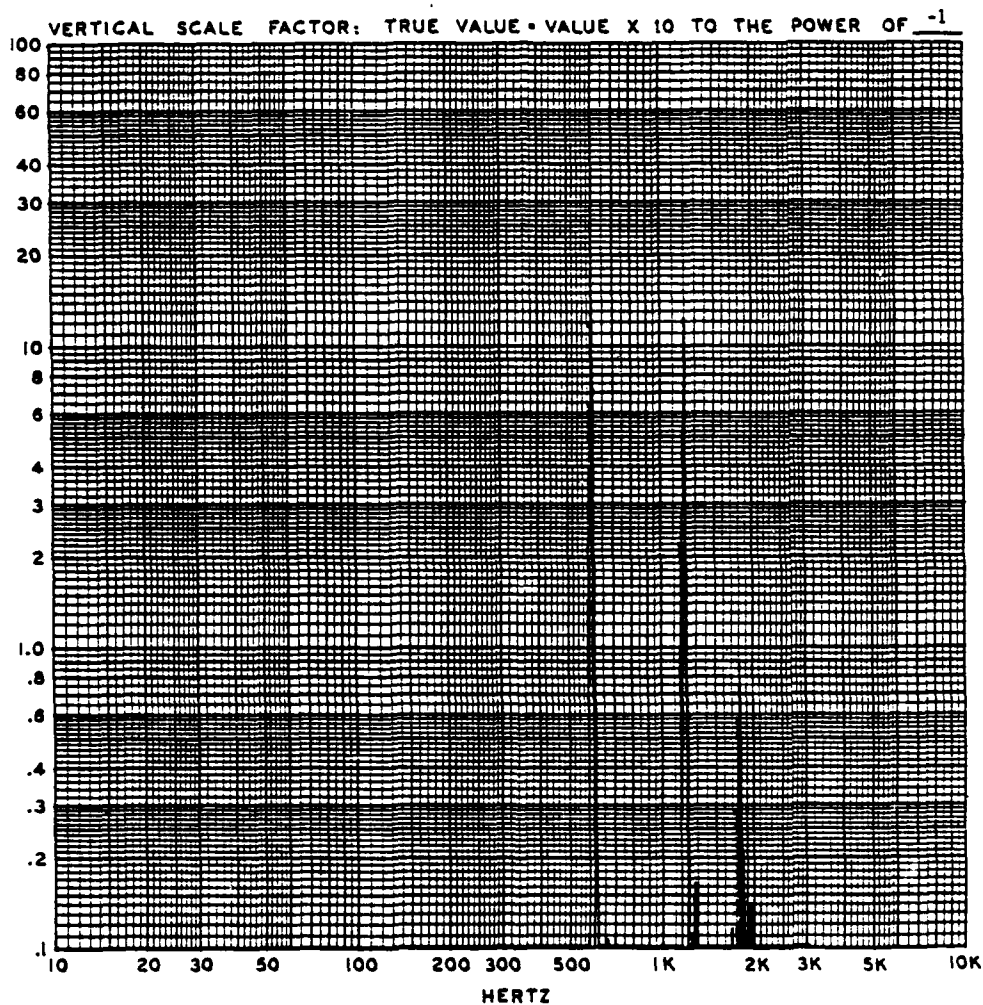
RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

PARAMETER <u>D0281A</u>	TEST DATE <u>15 September 81</u>
OVERALL RMS VALUE <u>2.9631 G's</u>	ANALYSIS BW <u>2.2467 Hz</u>
ENGINEERING UNITS <u>30.0000 G's</u>	ANALOG L.R. FILTER BW <u>2300.0000 Hz</u>
START TIME: (HR: MIN: SEC) <u>258/12:01:22</u>	DEGREES OF FREEDOM <u>9.0000</u>
SAMPLE RATE (S/SEC) <u>6134.9687</u>	PROCESS DATE <u>18 September 81</u>



RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

PARAMETER <u>00281A</u>	TEST DATE <u>15 September 81</u>
OVERALL RMS VALUE <u>6.4787 G's</u>	ANALYSIS BW <u>2.2467 Hz</u>
ENGINEERING UNITS <u>30.0000 G's</u>	ANALOG L.P. FILTER BW <u>2300.0000 Hz</u>
START TIME: (HR: MIN: SEC) <u>258/11:56:35</u>	DEGREES OF FREEDOM <u>9.0000</u>
SAMPLE RATE (S/SEC) <u>6134.9687</u>	PROCESS DATE <u>18 September 81</u>



RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

PARAMETER 00281 A

TEST DATE 15 September 81

OVERALL RMS VALUE 3.6288 G's

ANALYSIS BW 2.2467 Hz

ENGINEERING UNITS 30.0000 G's

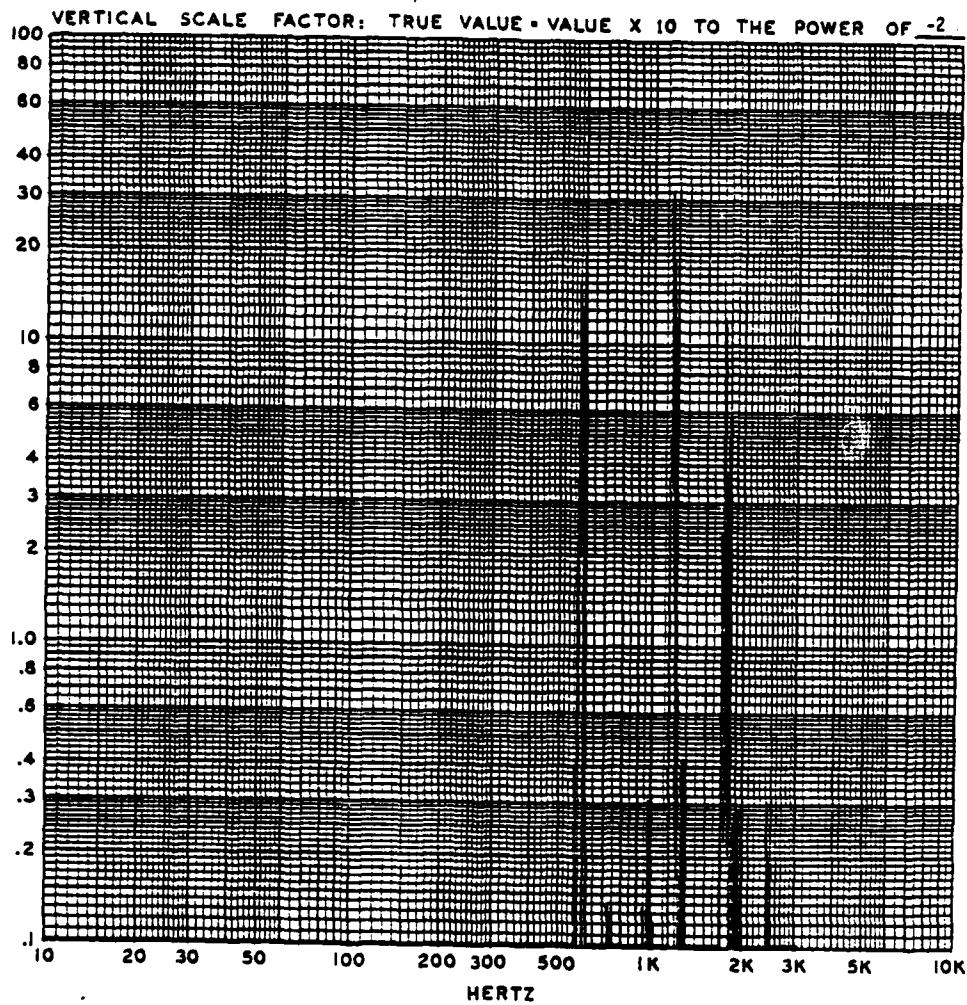
ANALOG L.P. FILTER BW 2300.0000 Hz

START TIME: (HR: MIN: SEC) 258/11:58:14

DEGREES OF FREEDOM 9.0000

SAMPLE RATE (S/SEC) 6134.9687

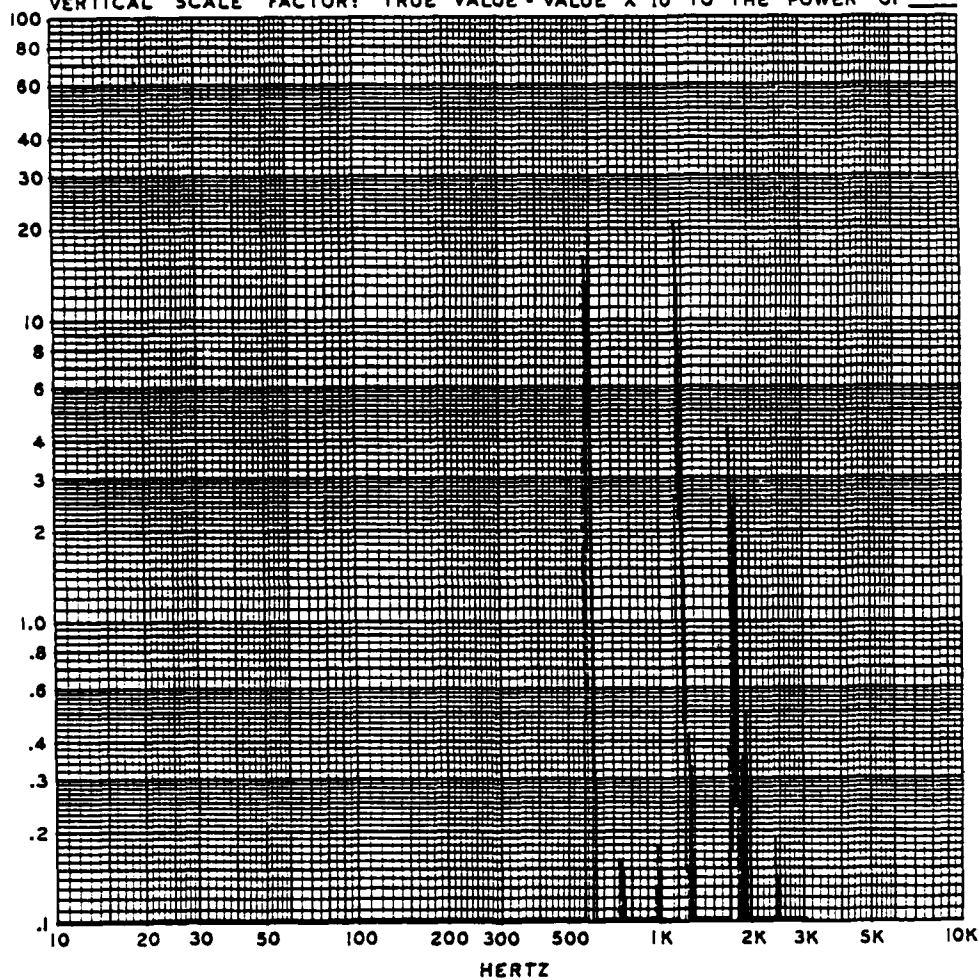
PROCESS DATE 18 September 81



RUN IDENTIFICATION APU HOTFIRE CHANNEL 10 STS-2

PARAMETER <u>00281A</u>	TEST DATE <u>15 September 81</u>
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ENGINEERING UNITS <u>30.0000 G's</u>	ANALOG L.P. FILTER B W <u>2300.0000 Hz</u>
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SAMPLE RATE (S/SEC) <u>6134.9687</u>	PROCESS DATE <u>18 September 81</u>

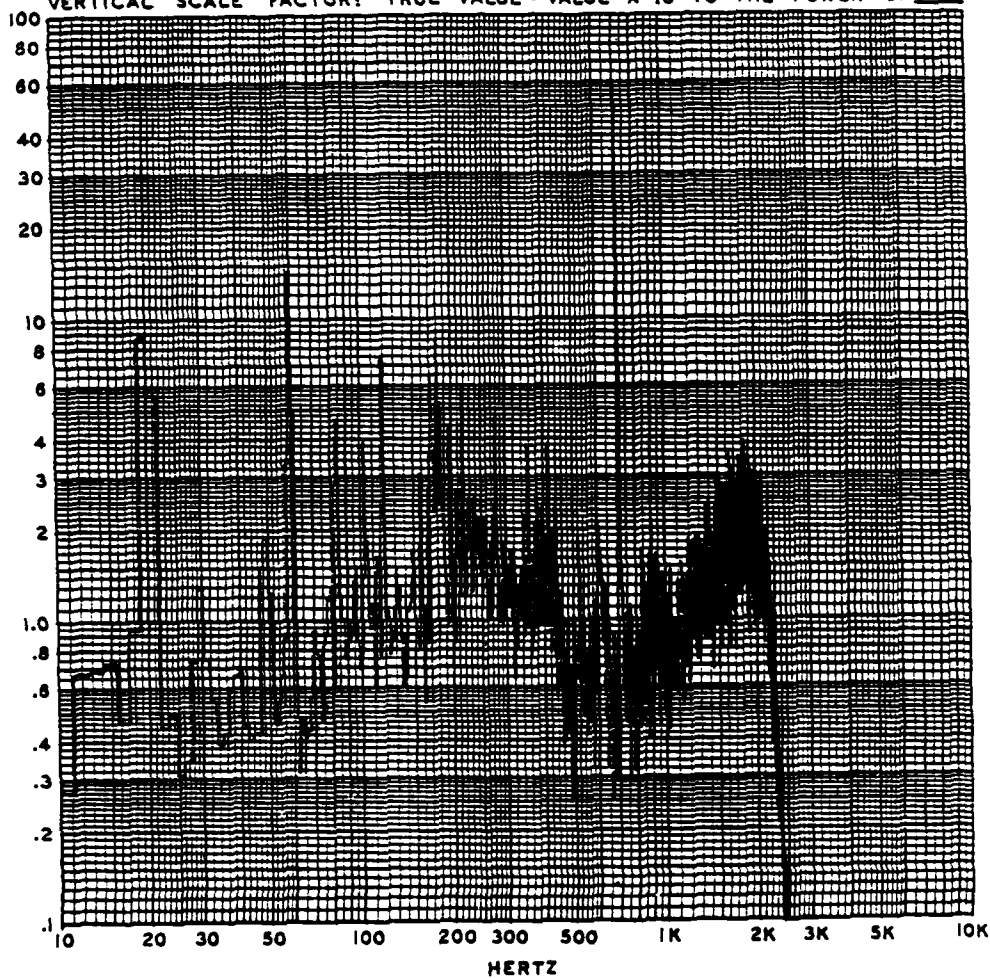
VERTICAL SCALE FACTOR: TRUE VALUE = VALUE X 10 TO THE POWER OF -2



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

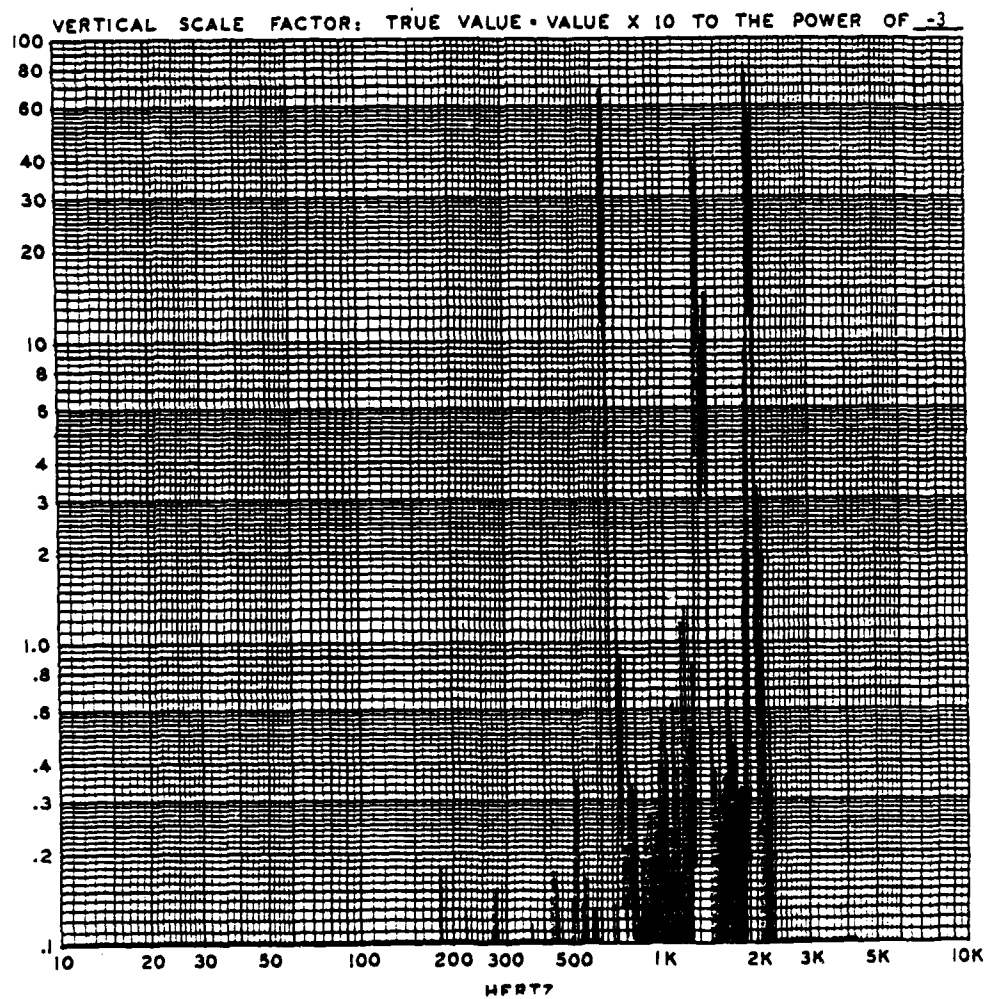
PARAMETER	D0280A	Noise Floor	TEST DATE	15 September 81
OVERALL RMS VALUE	0.1920 G's		ANALYSIS BW	2.2467 Hz
ENGINEERING UNITS	30.0000 Gs		ANALOG L.R FILTER BW	2300.0000 Hz
START TIME: (HR: MIN: SEC)	258/12:04:20		DEGREES OF FREEDOM	9.0000
SAMPLE RATE (S/SEC)	6134.9687		PROCESS DATE	18 September 81

VERTICAL SCALE FACTOR: TRUE VALUE = VALUE X 10 TO THE POWER OF -5



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

PARAMETER	<u>00280A</u>	TEST DATE	<u>15 September 81</u>
OVERALL RMS VALUE	<u>2.7269 G's</u>	ANALYSIS BW	<u>2.2467 Hz</u>
ENGINEERING UNITS	<u>30.0000 G's</u>	ANALOG L.P. FILTER BW	<u>2300.0000 Hz</u>
START TIME: (HR: MIN: SEC)	<u>258/12:03:20</u>	DEGREES OF FREEDOM	<u>9.0000</u>
SAMPLE RATE (S/SEC)	<u>6134.9687</u>	PROCESS DATE	<u>18 September 81</u>



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

PARAMETER D0280A

TEST DATE 15 September 81

OVERALL RMS VALUE 2.9619 G's

ANALYSIS BW 2.2467 Hz

ENGINEERING UNITS 30.0000 G's

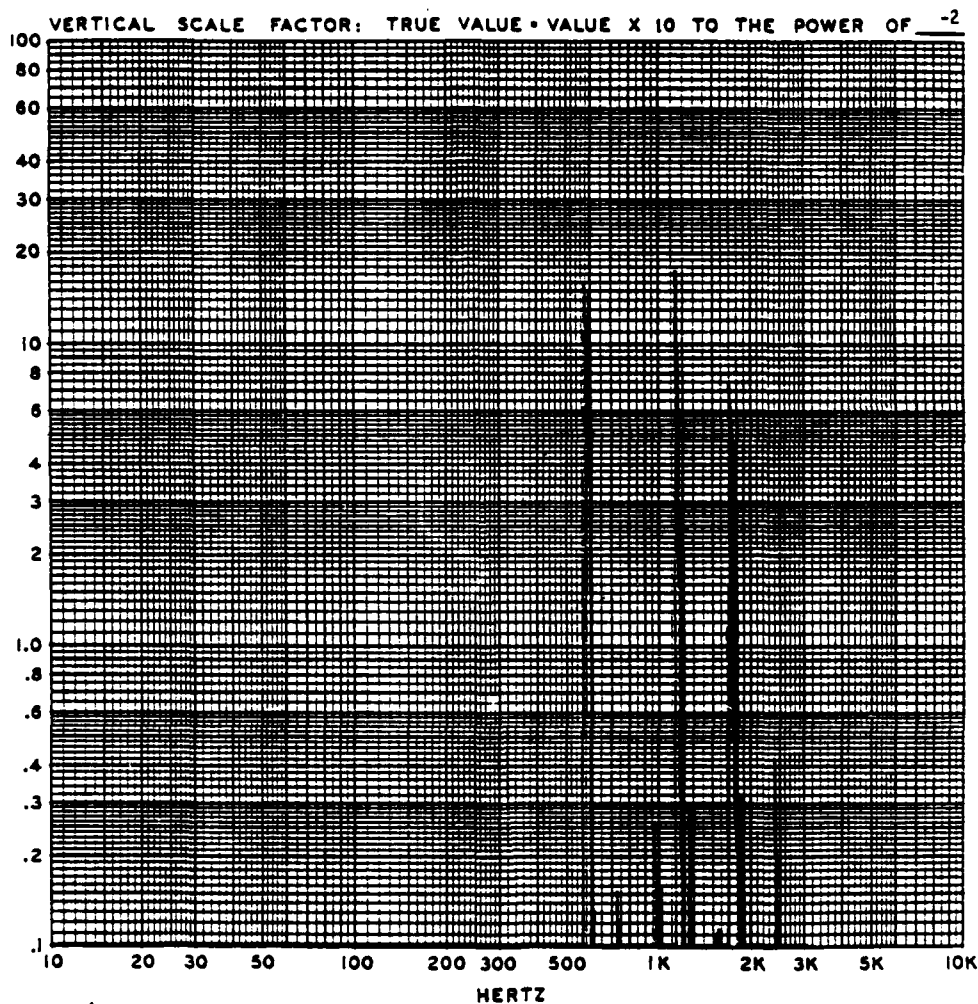
ANALOG L.R. FILTER BW 2300.0000 Hz

START TIME: (HR: MIN: SEC) 258/12:01:22

DEGREES OF FREEDOM 9.0000

SAMPLE RATE (S/SEC) 6134.9687

PROCESS DATE 18 September 81



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

PARAMETER D0280A

TEST DATE 15 September 81

OVERALL RMS VALUE 3.6335 G's

ANALYSIS BW 2.2467 Hz

ENGINEERING UNITS 30.0000 G's

ANALOG L.R. FILTER BW 2300.0000 Hz

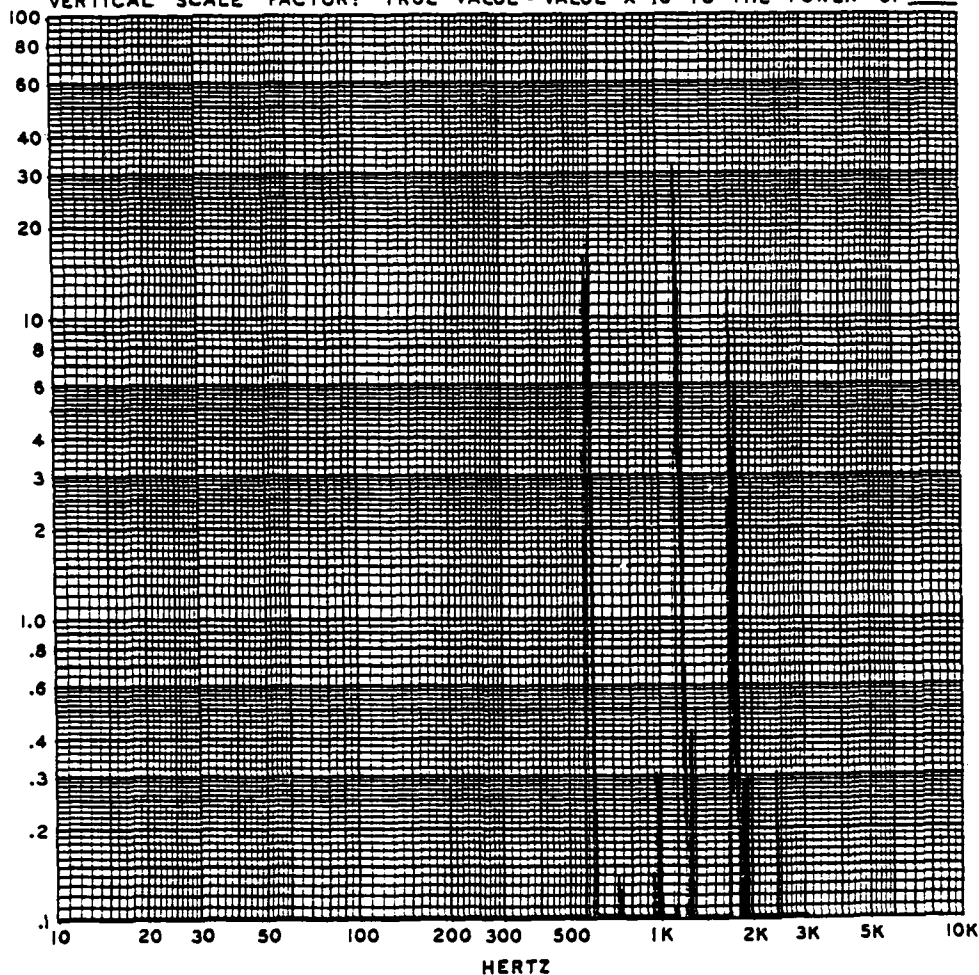
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DEGREES OF FREEDOM 9.0000

SAMPLE RATE (S/SEC) 6134.9687

PROCESS DATE 18 September 81

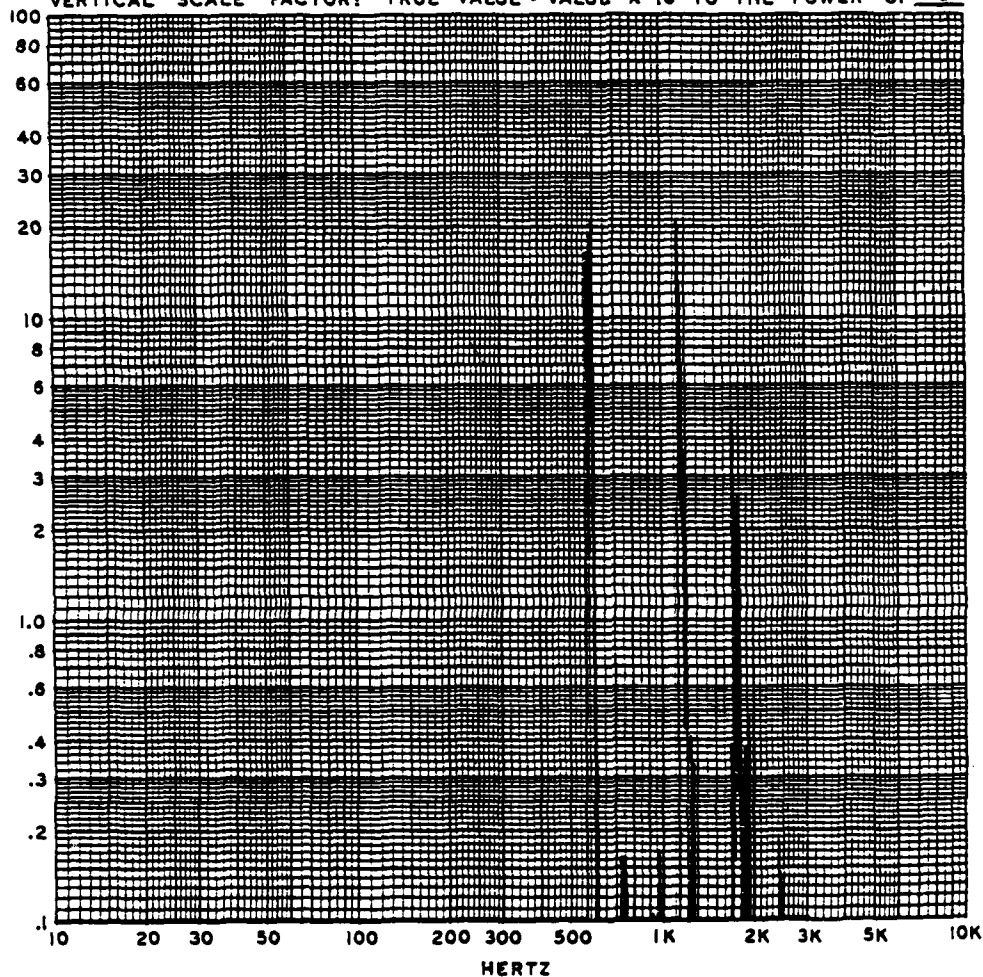
VERTICAL SCALE FACTOR: TRUE VALUE = VALUE X 10 TO THE POWER OF -2



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

PARAMETER	<u>00280A</u>	TEST DATE	<u>15 September 81</u>
OVERALL RMS VALUE	<u>3.1781 G's</u>	ANALYSIS BW	<u>2.2467 Hz</u>
ENGINEERING UNITS	<u>30.0000 G's</u>	ANALOG L.P. FILTER BW	<u>2300.0000 Hz</u>
START TIME: (HR: MIN: SEC)	<u>258/11:57:46</u>	DEGREES OF FREEDOM	<u>9.0000</u>
SAMPLE RATE (S/SEC)	<u>6134.9687</u>	PROCESS DATE	<u>18 September 81</u>

VERTICAL SCALE FACTOR: TRUE VALUE • VALUE X 10 TO THE POWER OF -2



RUN IDENTIFICATION APU HOTFIRE CHANNEL 11 STS-2

PARAMETER D0280A

TEST DATE 15 September 81

OVERALL RMS VALUE 6.4863 G's

ANALYSIS B W 2.2467 Hz

ENGINEERING UNITS 30.0000 G's

ANALOG L.P. FILTER B W 2300.0000 Hz

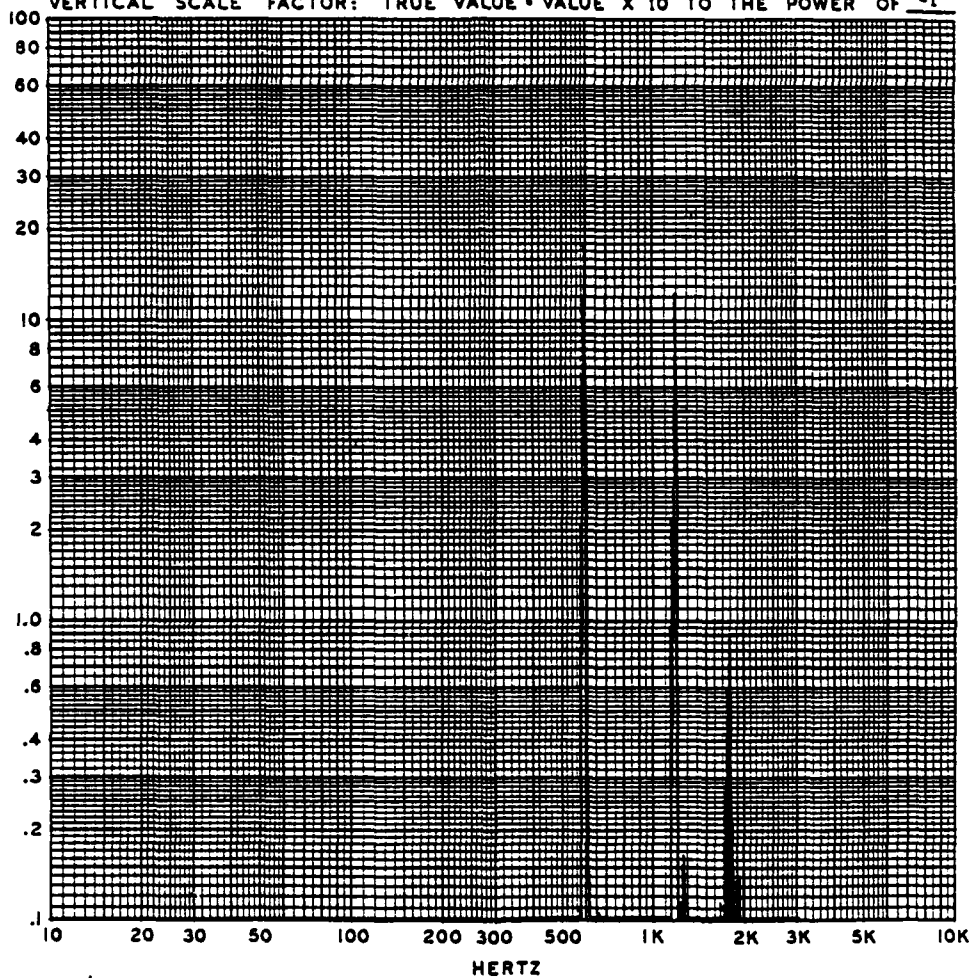
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DEGREES OF FREEDOM 9.0000

SAMPLE RATE (S/SEC) 6134.9687

PROCESS DATE 18 September 81

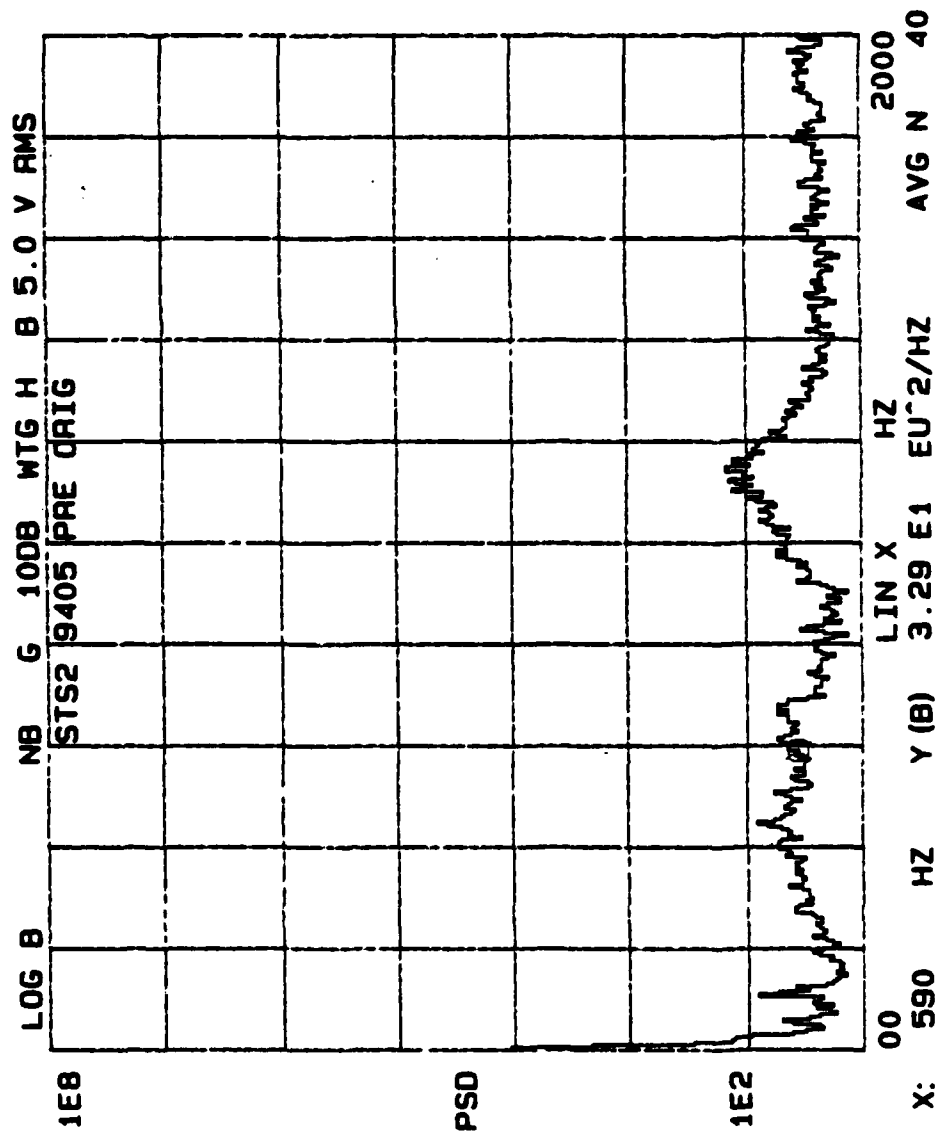
VERTICAL SCALE FACTOR: TRUE VALUE • VALUE X 10 TO THE POWER OF -1

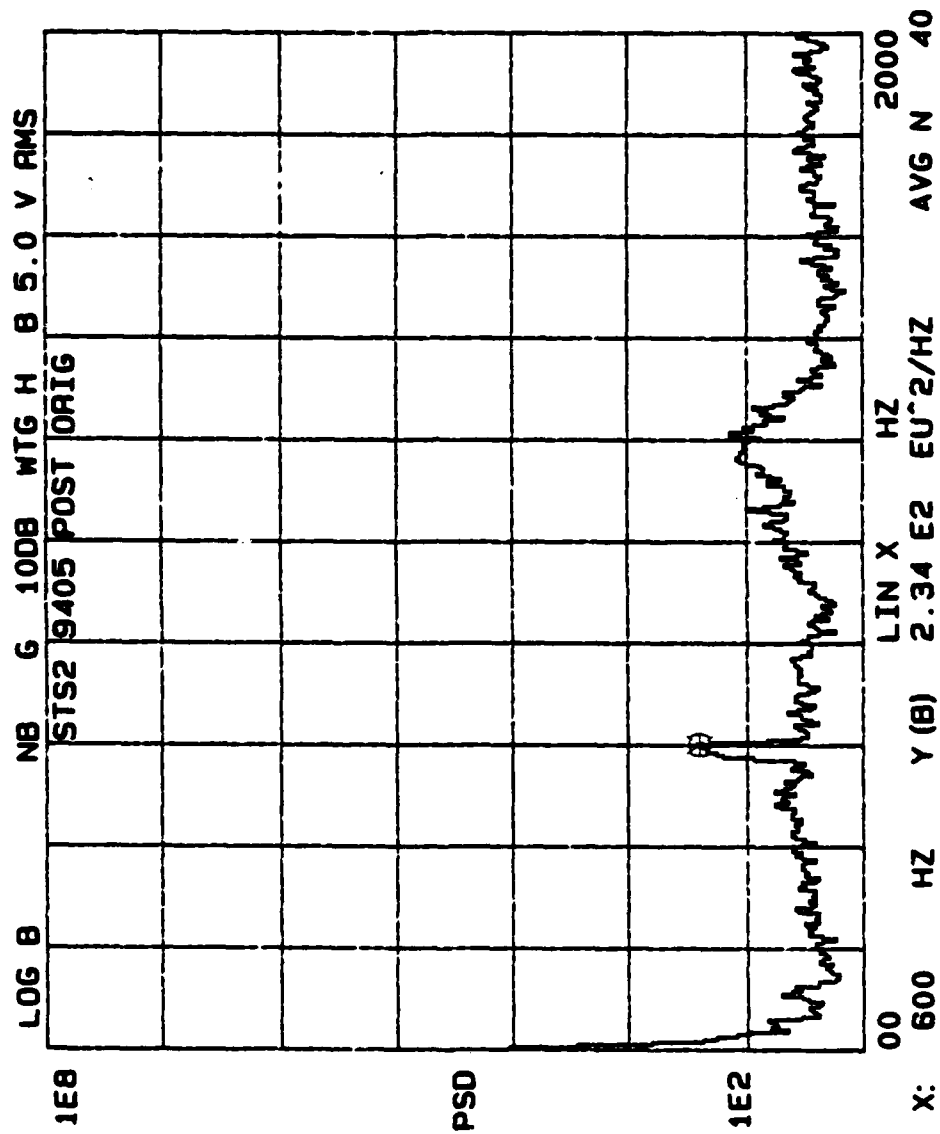


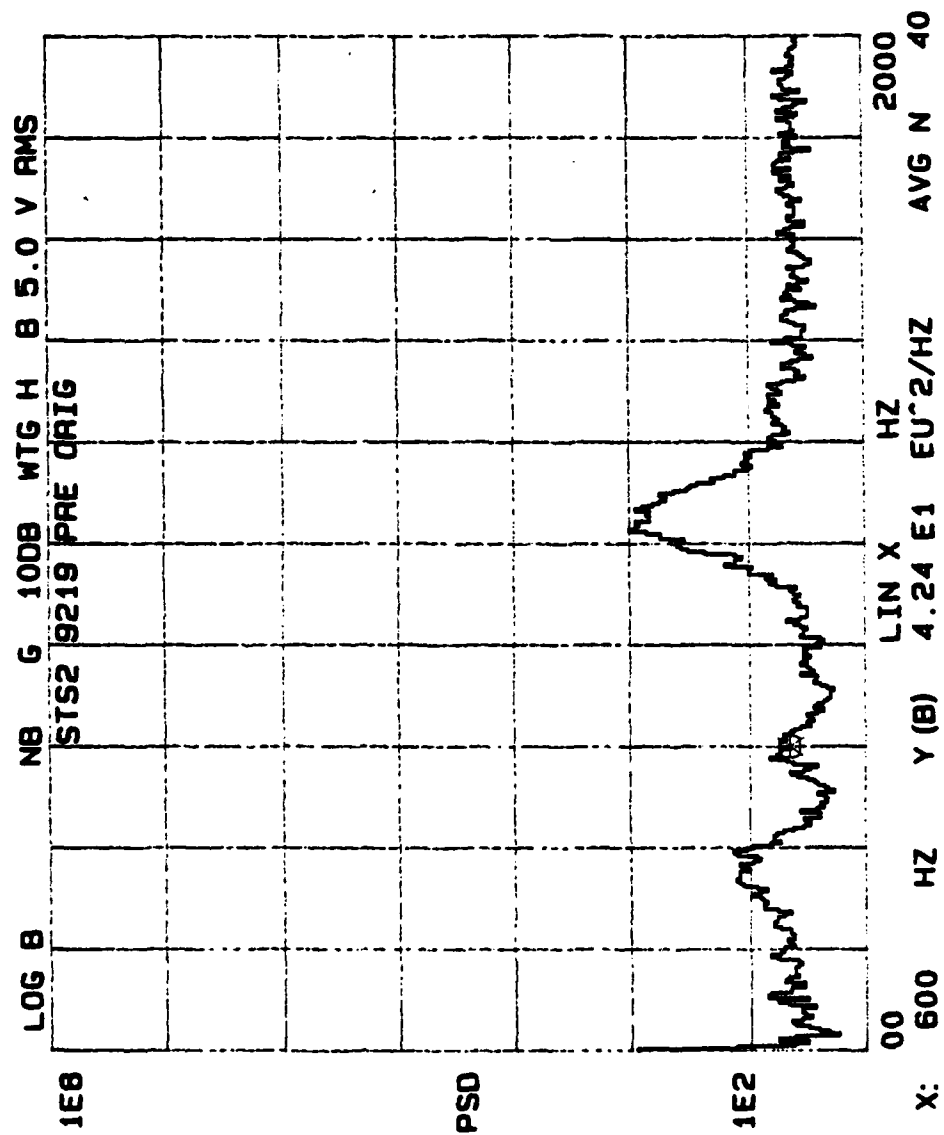
APPENDIX A

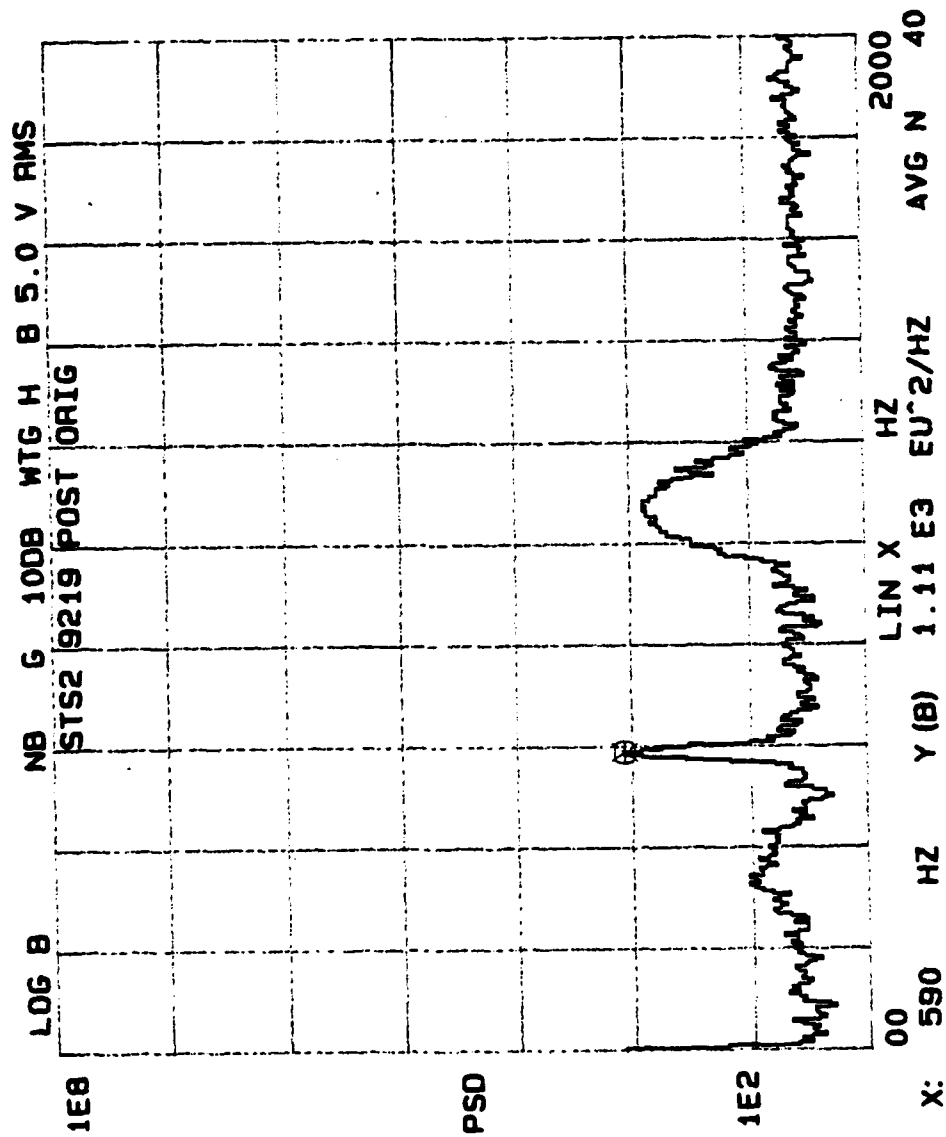
QUICK LOOK DATA PACKET

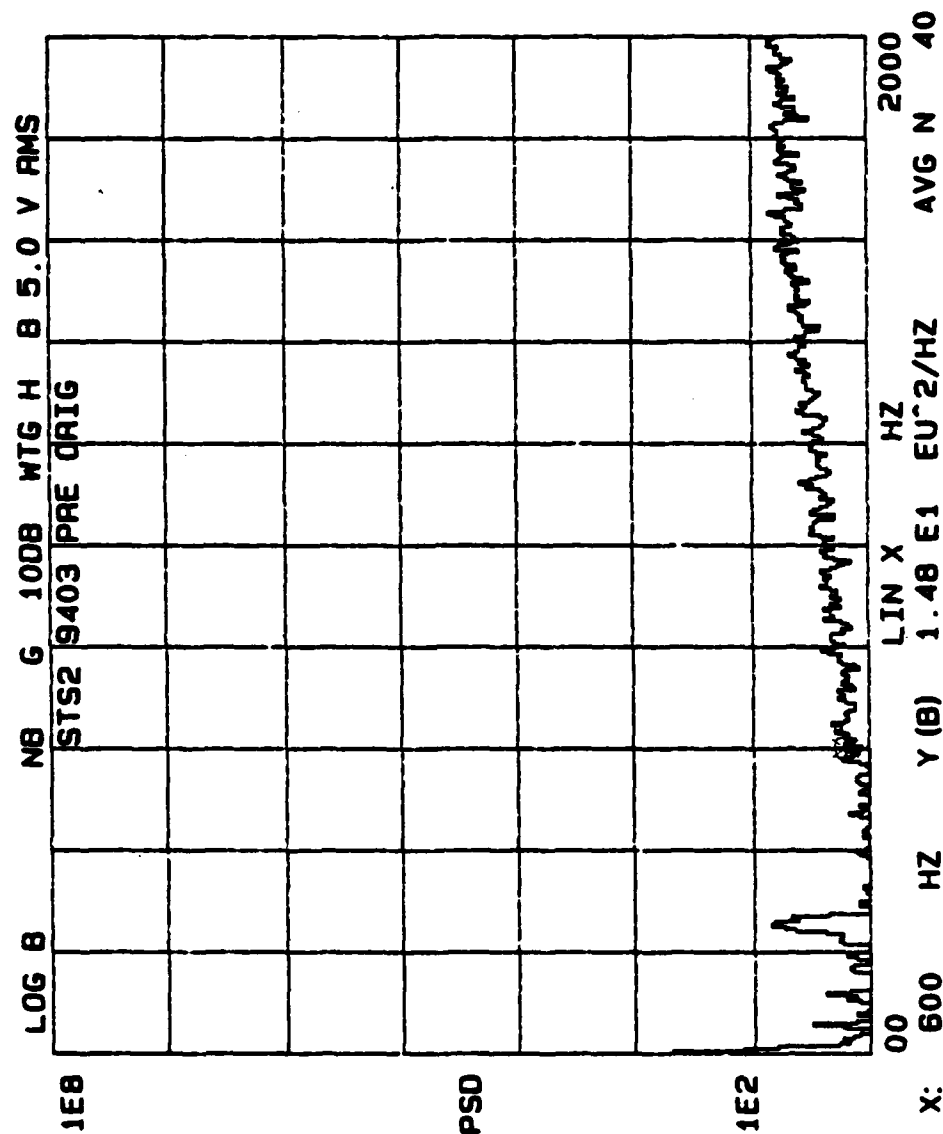
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TEST DATE 15 SEPTEMBER 81
RUN NO. (S)
REDUCTION DATE 18 SEPTEMBER 81
FACILITY JSC NASA VAFB BLDG #49
SEQUENCE NO. (S)
DATA TYPE PSD PLOTS ^{APU 2 x-axis APU 2 B} D0280A and D0281A + Noise Floor

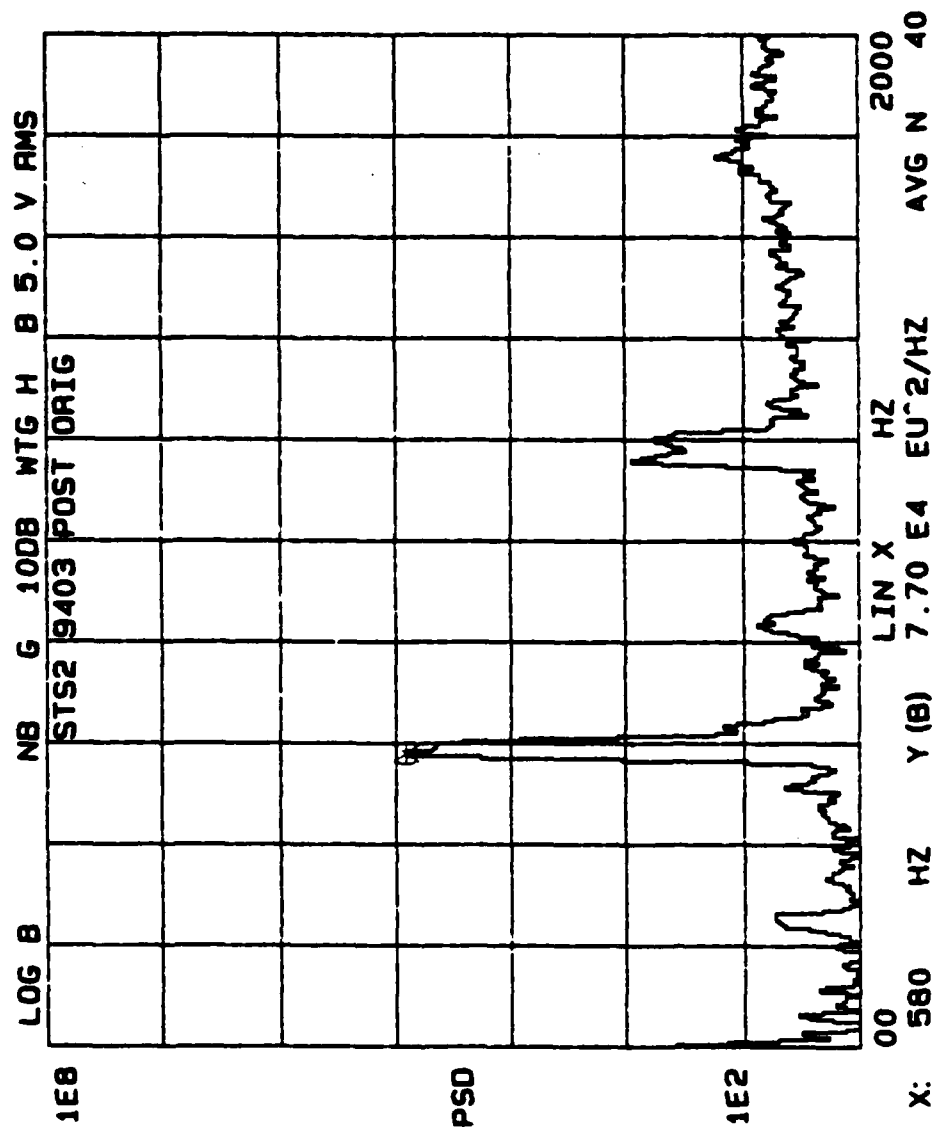


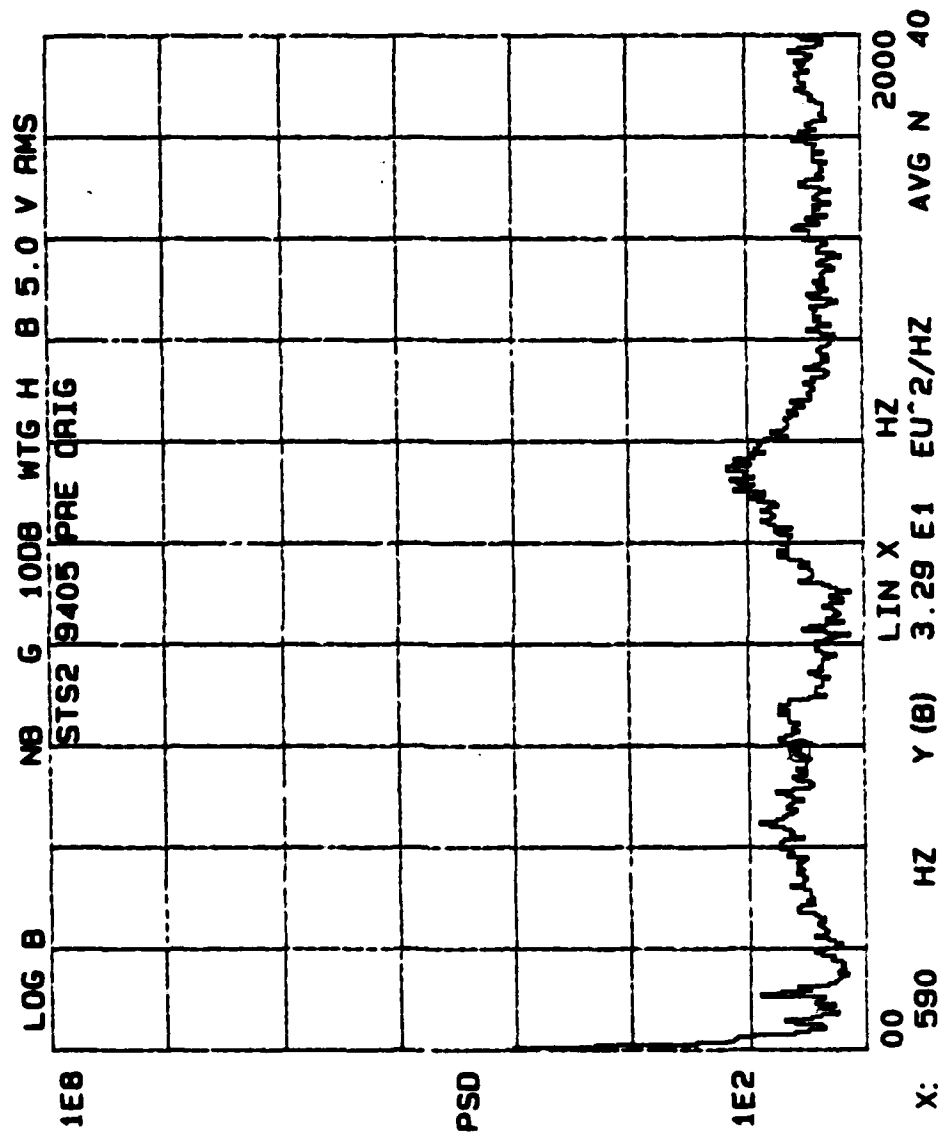


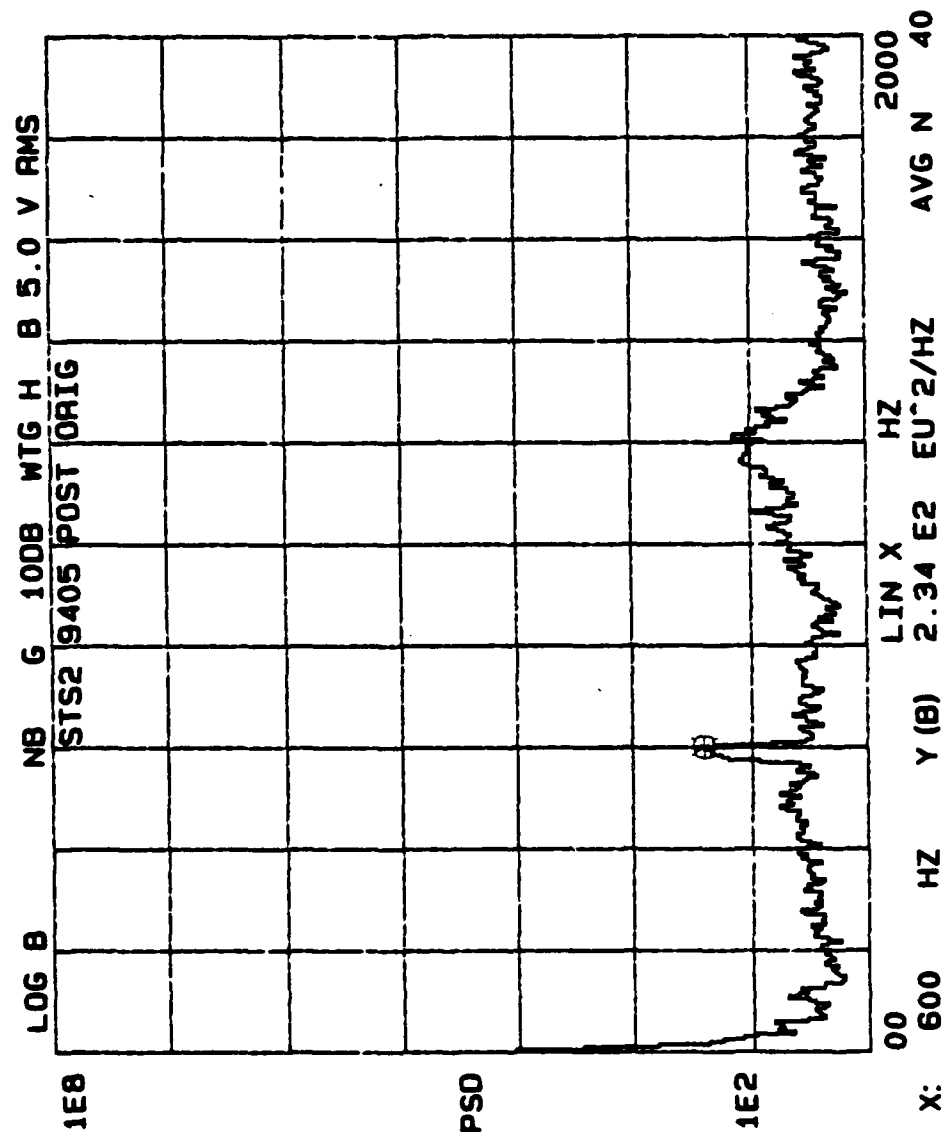


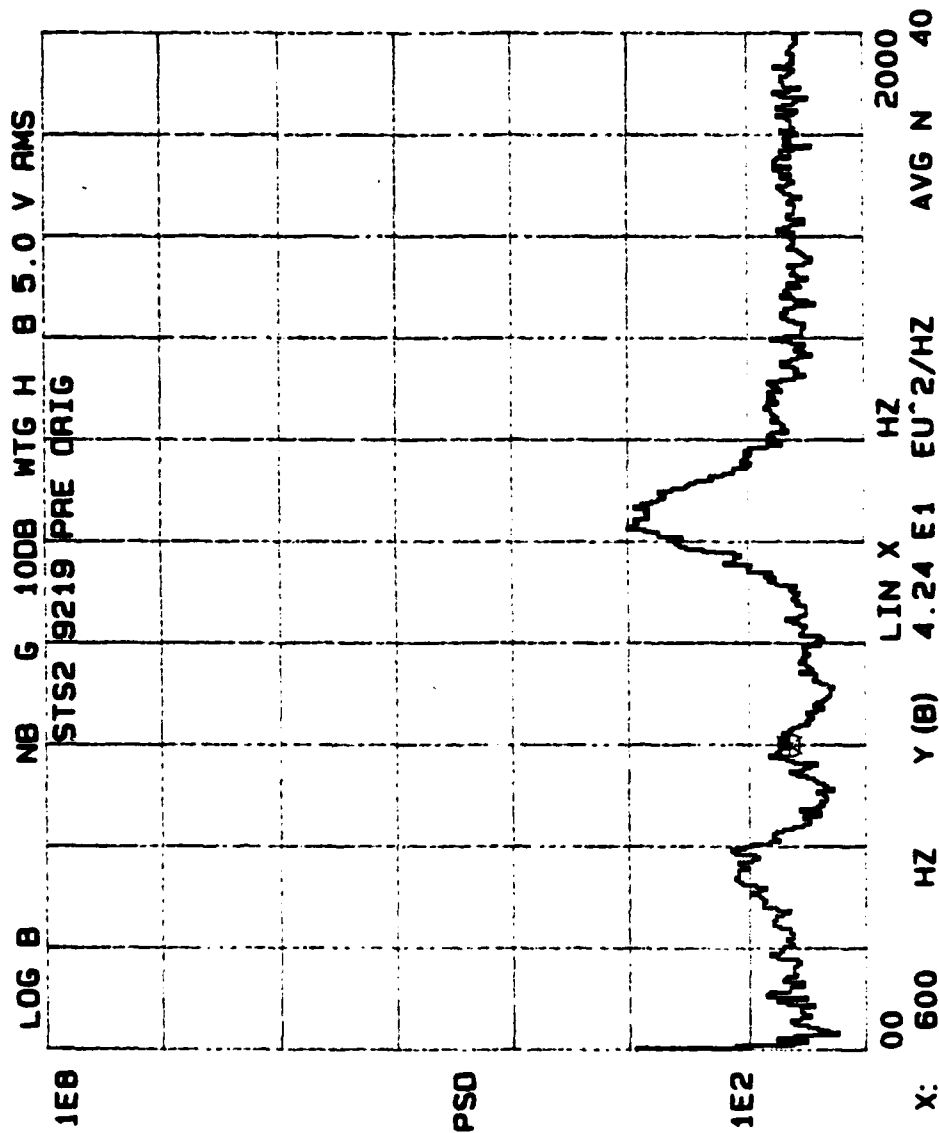


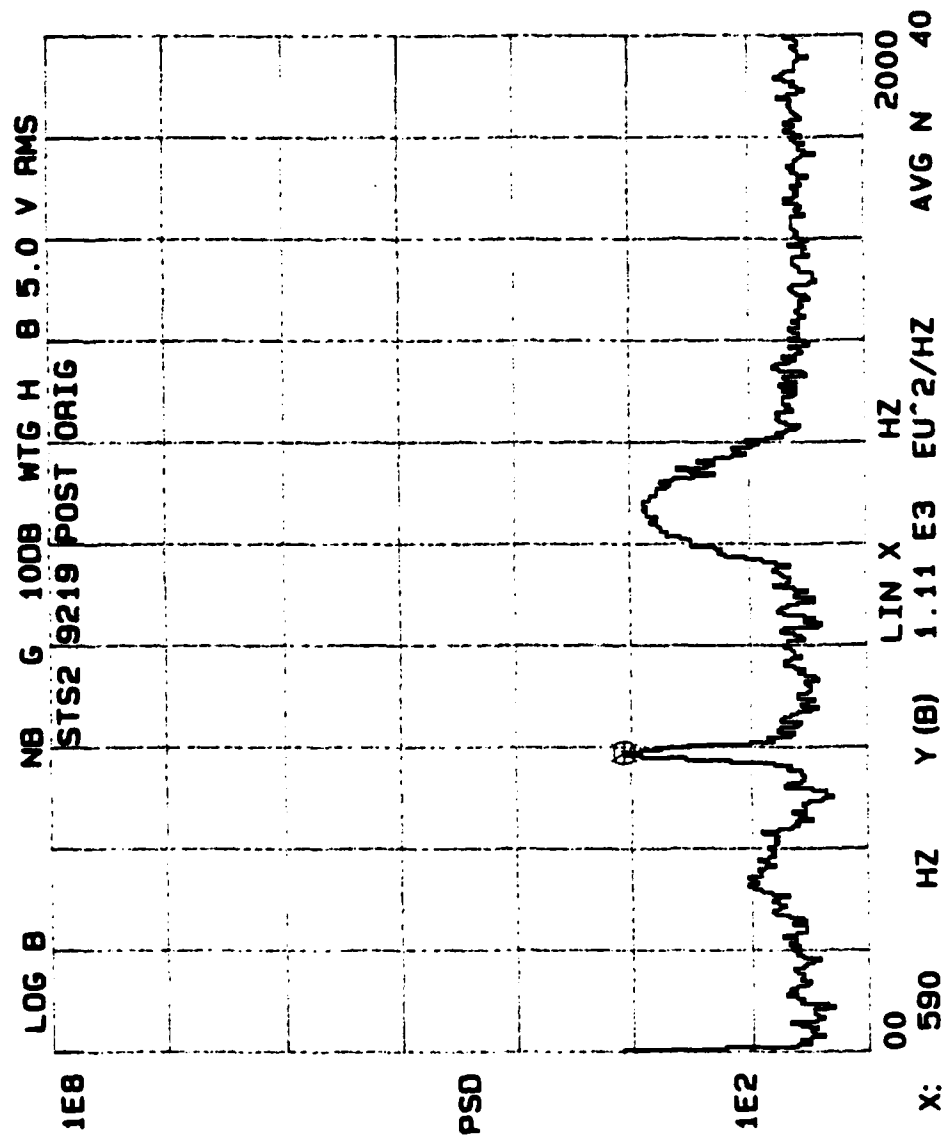


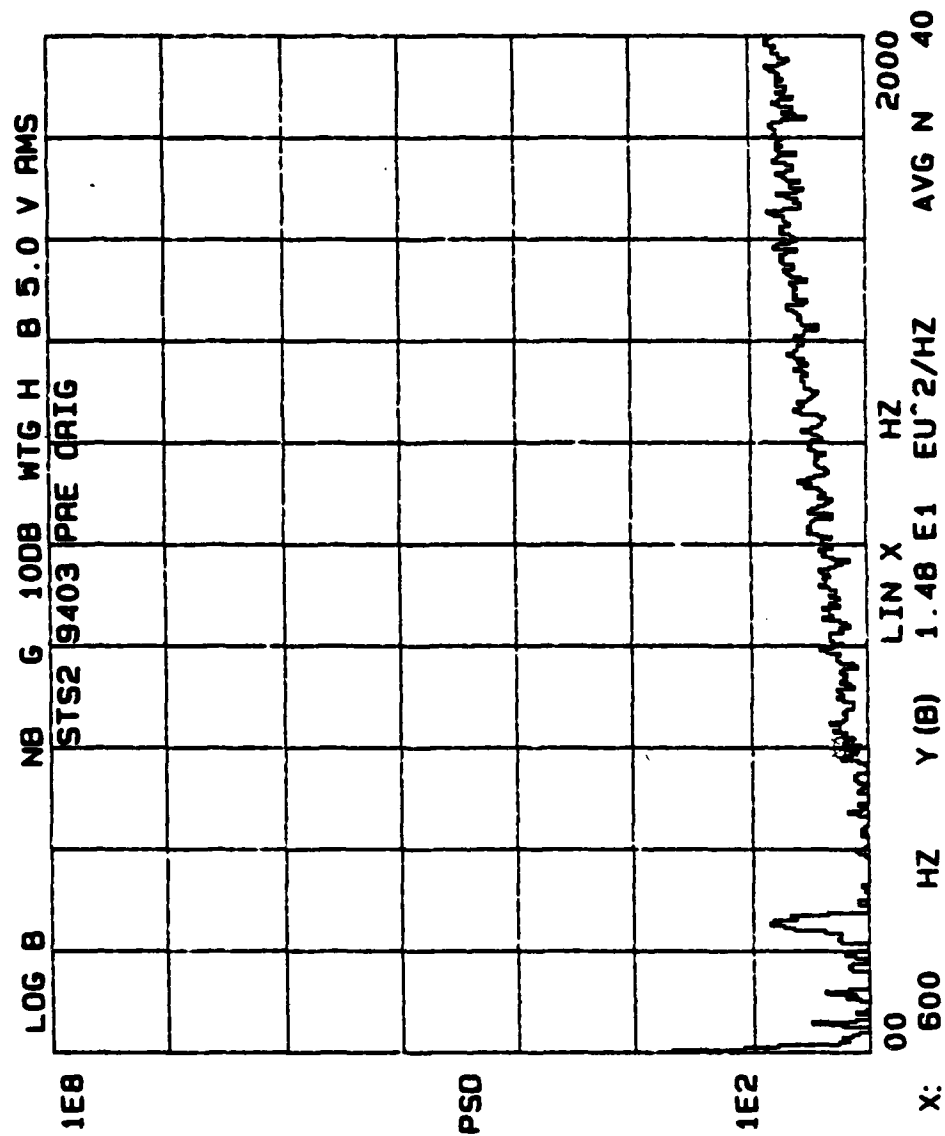


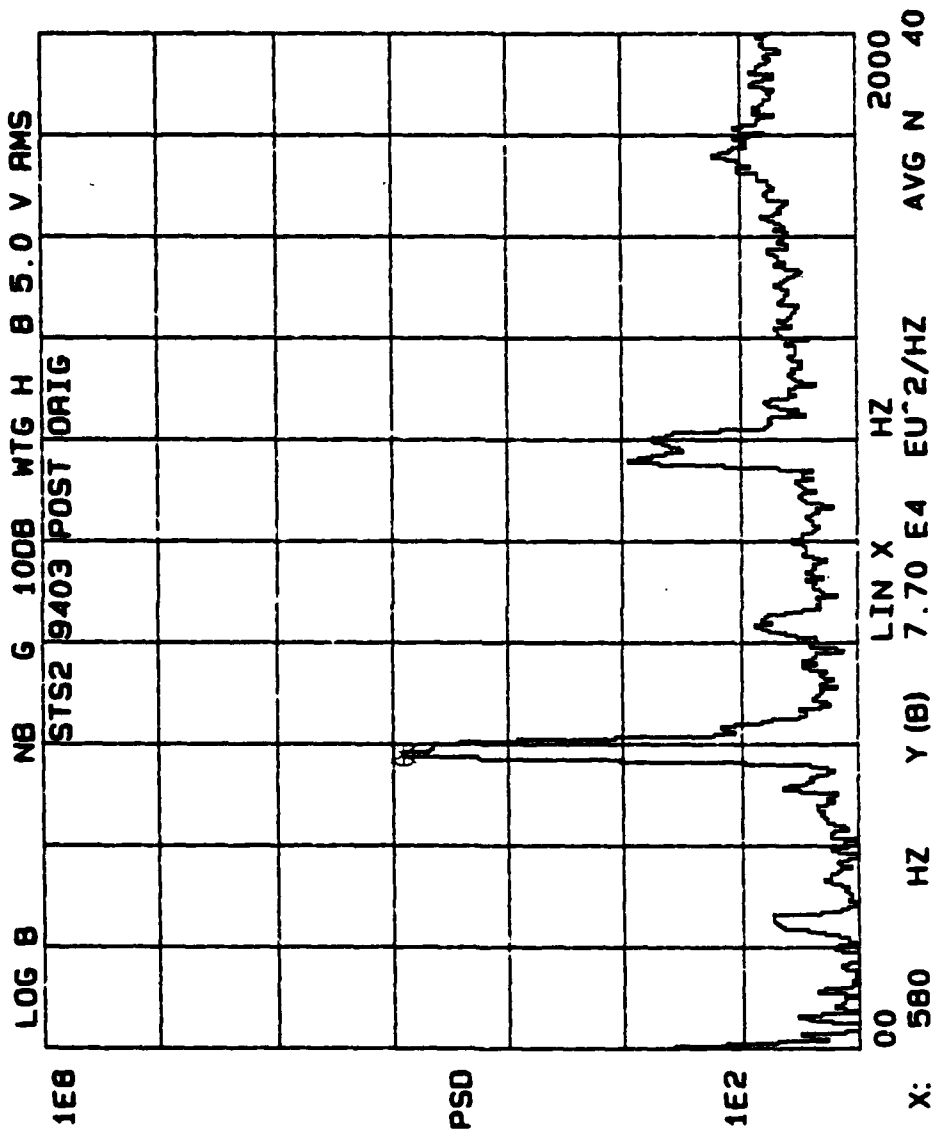


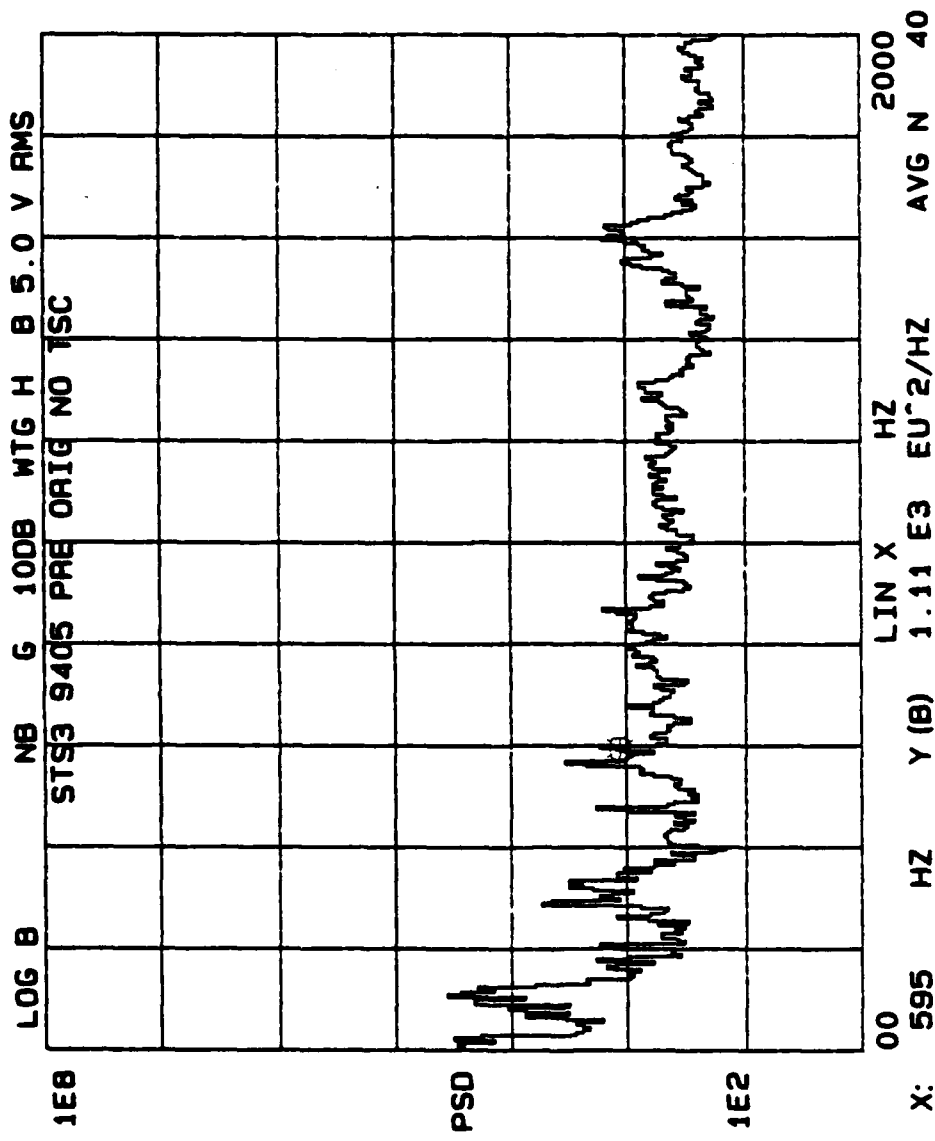


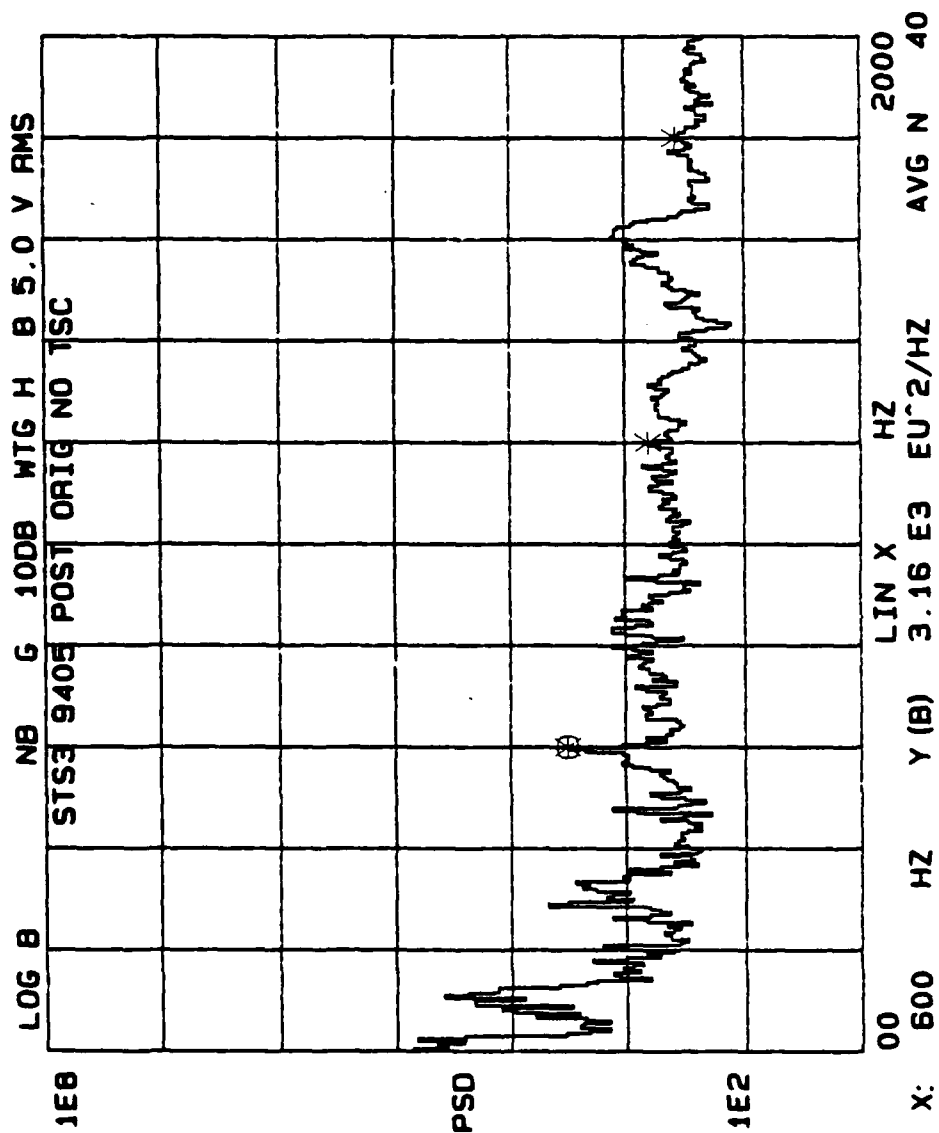













```

ALFA1=A1
ALFA2=1./A1
BETA1=D/E
BETA2=1./(D*E)
GAMMA1=D**2.
GAMMA2=1./D**2.
NUM1(1)=ALFA1*(WODIG**2.)
NUM1(2)=0.
NUM1(3)=1.
DEN1(1)=GAMMA1*(WODIG**2.)
DEN1(2)=BETA1*WODIG
DEN1(3)=1.
NUM2(1)=ALFA2*(WODIG**2.)
NUM2(2)=0.
NUM2(3)=1.
DEN2(1)=GAMMA2*(WODIG**2.)
DEN2(2)=BETA2*WODIG
DEN2(3)=1.
RHO=RHO1*RHO2
INUM1=3
INUM2=3
IDEN1=3
IDEN2=3
CALL PMPY(ANUM,IANUM,NUM1,INUM1,NUM2,INUM2)
CALL PMPY(ADEN,IADEN,DEN1,IDEN1,DEN2,IDEN2)
WRITE(4,110)RHO
WRITE(4,120)ANUM(5),ANUM(4),ANUM(3),ANUM(2),ANUM(1)
DO 105 I=1,5
  ANUM(I)=ANUM(I)*RHO
105 CONTINUE
WRITE(4,130)ANUM(5),ANUM(4),ANUM(3),ANUM(2),ANUM(1)
DO 106 I=1,5
  ANUM(I)=ANUM(I)/RHO
106 CONTINUE
WRITE(4,140)ADEN(5),ADEN(4),ADEN(3),ADEN(2),ADEN(1)
110 FORMAT('1','SECTION 2 OUTPUT',//,
&' ANALOG ELLIPTIC BANDPASS FILTER TRANSFER FUNCTION'
&,//,' NUMERATOR COEFFICIENT = RHO = ',F9.6,/)
120 FORMAT(' NUMERATOR POLYNOMIAL (NORMALIZED)'
&,/,E12.5,' S**4 +',/,E12.5,' S**3 +',
&,/,E12.5,' S**2 +',/,E12.5,' S +',/,E12.5,/)
130 FORMAT(' NUMERATOR POLYNOMIAL (UN-NORMALIZED)'
&,/,E12.5,' S**4 +',/,E12.5,' S**3 +',
&,/,E12.5,' S**2 +',/,E12.5,' S +',/,E12.5,/)
140 FORMAT(' DENOMINATOR POLYNOMIAL (NORMALIZED)'
&,/,E12.5,' S**4 +',/,E12.5,
&' S**3 +',/,E12.5,' S**2 +',/,E12.5,' S +',/,E12.5,/)
C*****ABP02130
C
C SECTION 2A
C
C CALCULATE THE COMPLEX POLES AND ZEROS OF THE ANALOG FILTER
C
C*****ABP02190
  IPDEG = 4

```

```

ABP01660
ABP01670
ABP01680
ABP01690
ABP01700
ABP01710
ABP01720
ABP01730
ABP01740
ABP01750
ABP01760
ABP01770
ABP01780
ABP01790
ABP01800
ABP01810
ABP01820
ABP01830
ABP01840
ABP01850
ABP01860
ABP01870
ABP01880
ABP01890
ABP01900
ABP01910
ABP01920
ABP01930
ABP01940
ABP01950
ABP01960
ABP01970
ABP01980
ABP01990
ABP02000
ABP02010
ABP02020
ABP02030
ABP02040
ABP02050
ABP02060
ABP02070
ABP02080
ABP02090
ABP02100
ABP02110
ABP02120
ABP02130
ABP02140
ABP02150
ABP02160
ABP02170
ABP02180
ABP02190
ABP02200

```

FILE: ABPDBP FORTRAN A1

```
C      CHOOSE R17=R27=1./(W0*C1)(APPROX)=26.5 KOHMS      *ABP01110
C      USE MEASURED VALUES FOR CALCULATIONS      *ABP01120
C      *ABP01130
C*****ABP01140
      C11=.00996E-06      ABP01150
      C12=.00995E-06      ABP01160
      C21=.01030E-06      ABP01170
      C22=.01034E-06      ABP01180
      R17=26.7E+03      ABP01190
      R27=26.7E+03      ABP01200
C*****ABP01210
C      ELLIPTIC BANDPASS FILTER STAGE RESISTOR VALUES      ABP01220
C*****ABP01230
      R11=(E/(K*D*W0*C11))*SQRT(A/C)      ABP01240
      R12=K*R11*SQRT(C/A)      ABP01250
      R13=1./(D*W0*C11)      ABP01260
      R14=(R17/K)*SQRT(A/C)      ABP01270
      R15=(D/(K*A1*W0*C12))*SQRT(A/C)      ABP01280
      R16=C11*R13/C12      ABP01290
      WRITE(4,20)C11,C12,R17,R11,R12,R13,R14,R15,R16      ABP01300
      R21=((D*E)/(K*W0*C21))*SQRT(A/C)      ABP01310
      R22=(K*R21)*SQRT(C/A)      ABP01320
      R23=D/(W0*C21)      ABP01330
      R24=(R27/K)*SQRT(A/C)      ABP01340
      R25=(A1/(K*D*W0*C22))*SQRT(A/C)      ABP01350
      R26=C21*R23/C22      ABP01360
      WRITE(4,30)C21,C22,R27,R21,R22,R23,R24,R25,R26      ABP01370
10  FORMAT(' SECTION 1 OUTPUT',//,      ABP01380
      &' INPUT AND DERIVED PARAMETERS FOR FURTHER CALCULATIONS'      ABP01390
      &',//',' A = ',F9.6/,', B = ',F9.6/,', C = ',F9.6/,', D = ',      ABP01400
      &F9.6/,', E = ',F9.6/,', A1 = ',F9.6/,', F0 = ',F9.3/,', W0 = '      ABP01410
      &',F9.3/,', Q = ',F9.6/,', K = ',F9.6/,', K1 = ',F9.6/,',      ABP01420
      &' K2 ',F9.6/,', W0(DIG) = ',F9.3/,', F0(DIG) = ',F9.3,///)      ABP01430
20  FORMAT(' ELLIPTIC ANALOG BPF COMPONENT VALUES',//,' FIRST STAGE'      ABP01440
      &',//',' C11 = ',E8.3/,', C12 = ',E8.3/,', R17 = ',E8.3,      ABP01450
      &',//',' R11 = ',E8.3/,', R12 = ',E8.3/,', R13 = ',E8.3/,',      ABP01460
      &' R14 = ',E8.3/,', R15 = ',E8.3/,', R16 = ',E8.3,///)      ABP01470
30  FORMAT(' SECOND STAGE',//,      ABP01480
      &' C21 = ',E8.3/,', C22 = ',E8.3/,', R27 = ',E8.3,      ABP01490
      &',//',' R21 = ',E8.3/,', R22 = ',E8.3/,', R23 = ',E8.3/,',      ABP01500
      &' R24 = ',E8.3/,', R25 = ',E8.3/,', R26 = ',E8.3)      ABP01510
C*****ABP01520
C      *ABP01530
C      *ABP01540
C      *ABP01550
C      IN THIS PORTION OF THE PROGRAM WE COMPUTE THE ANALOG TRANSFER      *ABP01560
C      FUNCTION OF THE ELLIPTIC BAND PASS FILTER. IF THE ANALOG      *ABP01570
C      FUNCTION ALONE IS DESIRED THEN WE USE W0 FOR CALCULATIONS.      *ABP01580
C      IF THE ANALOG TRANSFER FUNCTION IS DESIRED FOR DIGITAL      *ABP01590
C      TRANSFORMATION THEN WE MUST USE THE PRE-WARPED ANALOG TO W0      *ABP01600
C      WHICH IS W0DIG.      ABP01610
C      *ABP01620
C*****ABP01630
      RH01=K1*SQRT(C/A)      ABP01640
      RH02=K2*SQRT(C/A)      ABP01650
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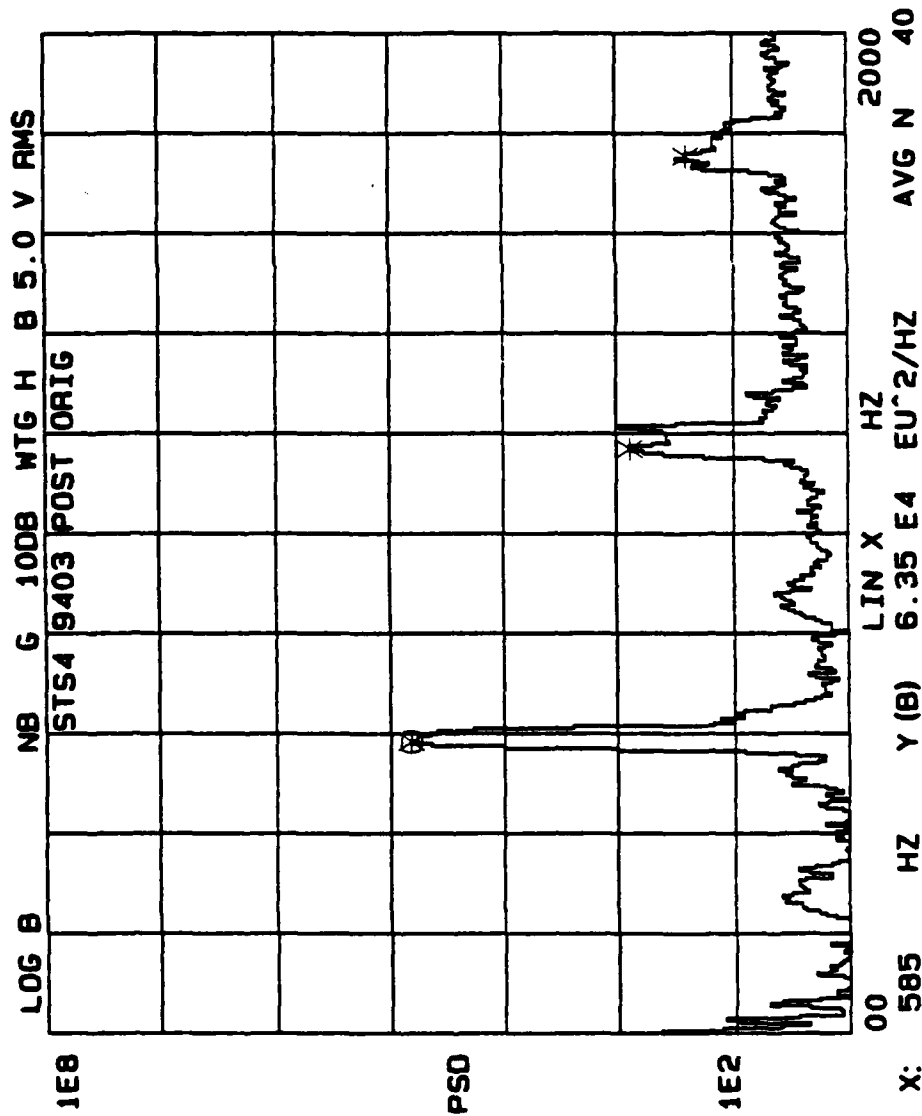
      COMPLEX AZERO(4),APOLE(4)
C SECTION 3
      INTEGER IZM,IZP,IDNUM,IDDEN,IDTMP
      REAL TNCOE,TDCOE
      REAL ZM1(5),ZM2(5),ZM3(5),ZM4(5),ZP1(5),ZP2(5),ZP3(5),ZP4(5)
      REAL DNUM(5),DDEN(5),DTMP(9)
C SECTION 3A
      REAL DNMINV(5),DDNMINV(5)
      REAL RZ(4),RP(4)
      COMPLEX DZERO(4),DPOLE(4)
C SECTION 4
      INTEGER I4,I2
      COMPLEX*16 DZ4(4),DP4(4),DZ21(2),DP21(2),DZ22(2),DP22(2)
      COMPLEX*16 DN4(4),DD4(4),DN21(2),DD21(2),DN22(2),DD22(2)
      COMPLEX DN45(5),DD45(5),DN213(3),DD213(3),DN223(3),DD223(3)
C*****
C TABULATED INPUT PARAMETERS FOR DESIRED SECOND ORDER LOWPASS
C FILTER EQUIVALENT HAVING THE FOLLOWING CHARACTERISTICS:
C N = 2
C MAXIMUM PASSBAND RIPPLE WIDTH (PRW) = 2.0 DB
C MINIMUM LOSS IN THE STOPBAND (MSL) = 30.0 DB
C NORMALIZED TRANSITION WIDTH = 2.2921
C FILTER GAIN (K) = 1.0
C*****
      F0=600.
      A=21.164003
      B=0.787152
      C=0.842554
      Q=12.
      P1=3.1415927
      K=1.
C*****
C DERIVED PARAMETERS
C*****
      E=(1./B)*SQRT((C+4.*Q**2.+SQRT((C+4.*Q**2.)*2.- (2.*B*Q)**2.))/2.)
      D=.5*((B*E/Q)+SQRT((B*E/Q)**2.-4.))
      A1=1.+(1./(2.*Q**2.))*(A+SQRT(A**2.+4.*A*Q**2.))
      K1=SQRT(K)
      K2=K1
      T=4.*192./6.666E+06
      TDIV2=T/2.
      W0=2.*PI*F0
      WODIG=(1./TDIV2)*ATAN(W0*TDIV2)
      FODIG=WODIG/(2.*PI)
      WRITE(4,10)A,B,C,D,E,A1,F0,W0,Q,K,K1,K2,WODIG,FODIG
C*****
C
C SECTION 1
C
C THIS PROGRAM SECTION COMPUTES ANALOG ELLIPTIC BANDPASS FILTER
C RESISTOR AND CAPACITOR VALUES USING THE ABSOLUTE AND DERIVED
C PARAMETERS CALCULATED ABOVE:
C
C CHOOSE C11=C21=.01E-06 (APPROX)=.01 UF
C CHOOSE C12=C22=.01E-06 (APPROX)=.01 UF

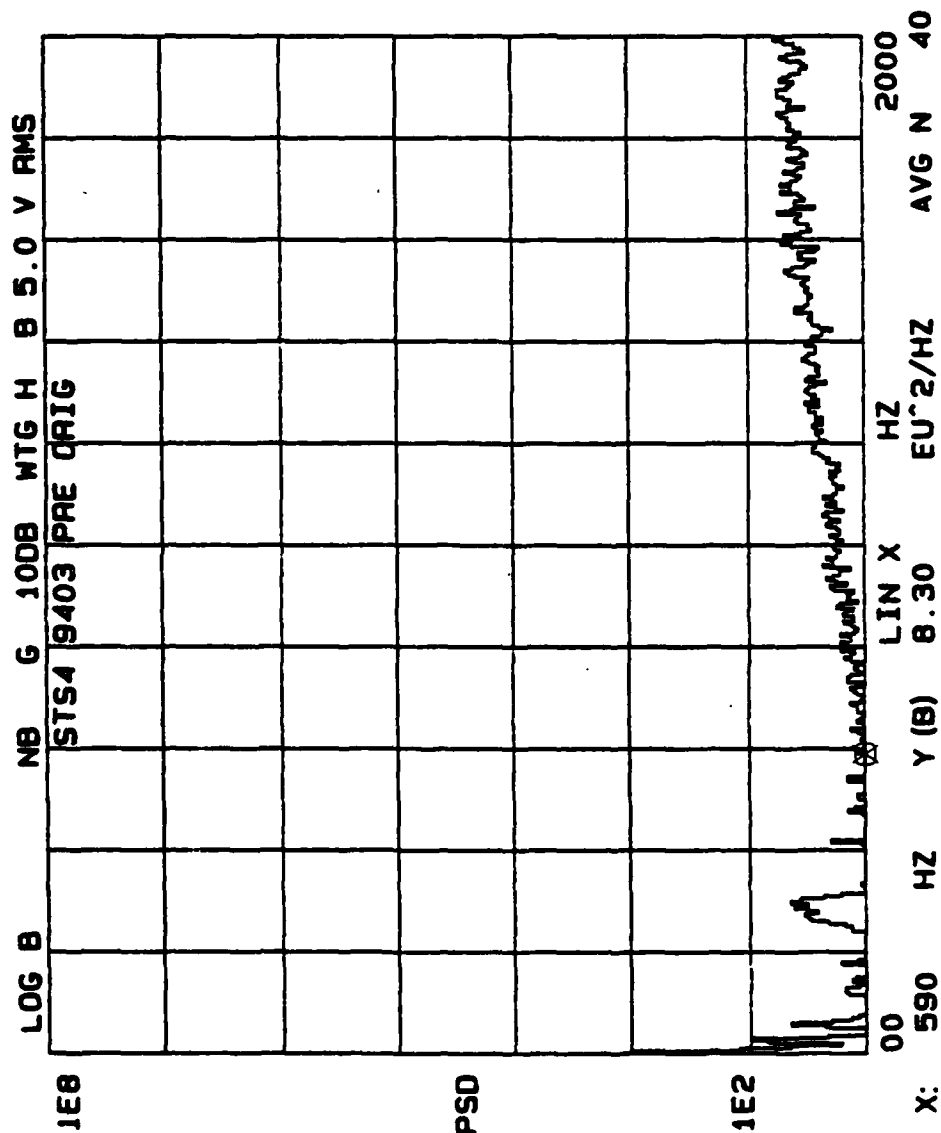
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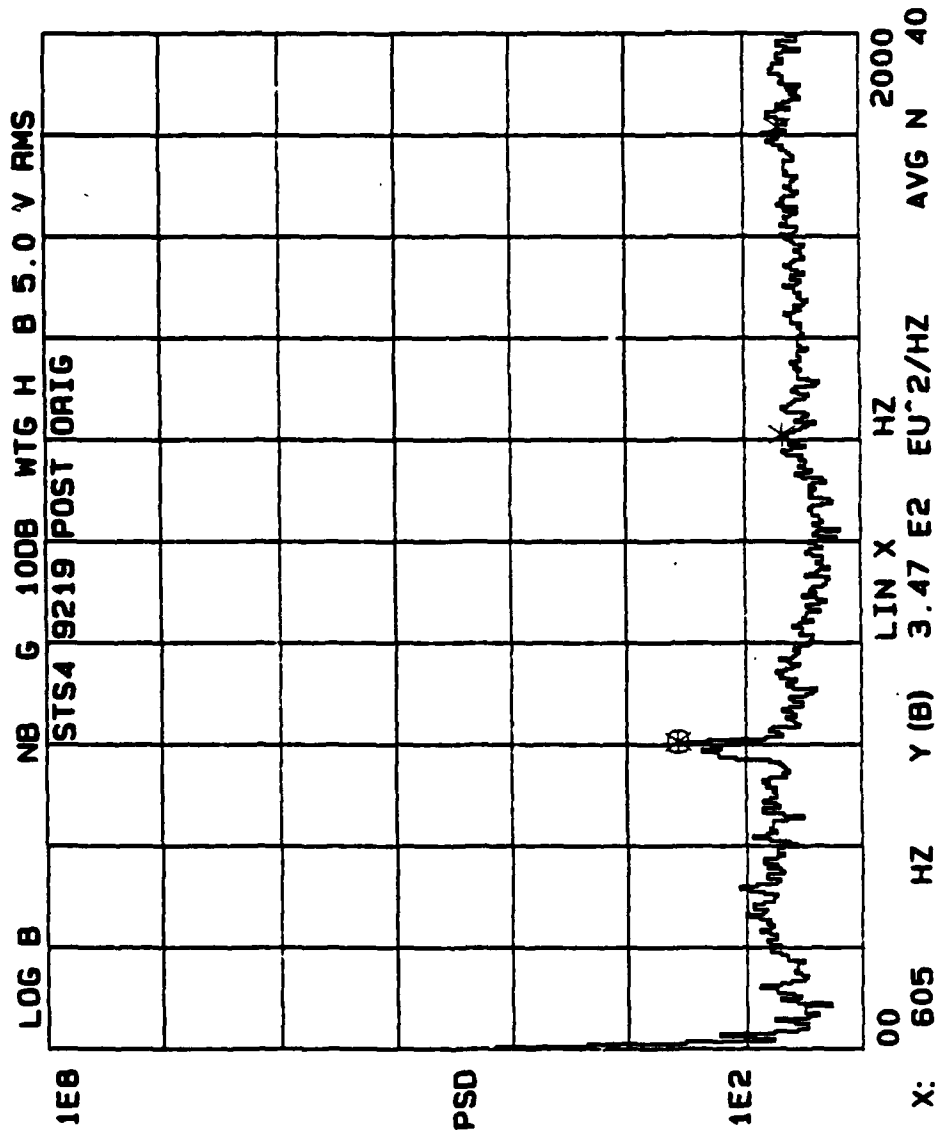
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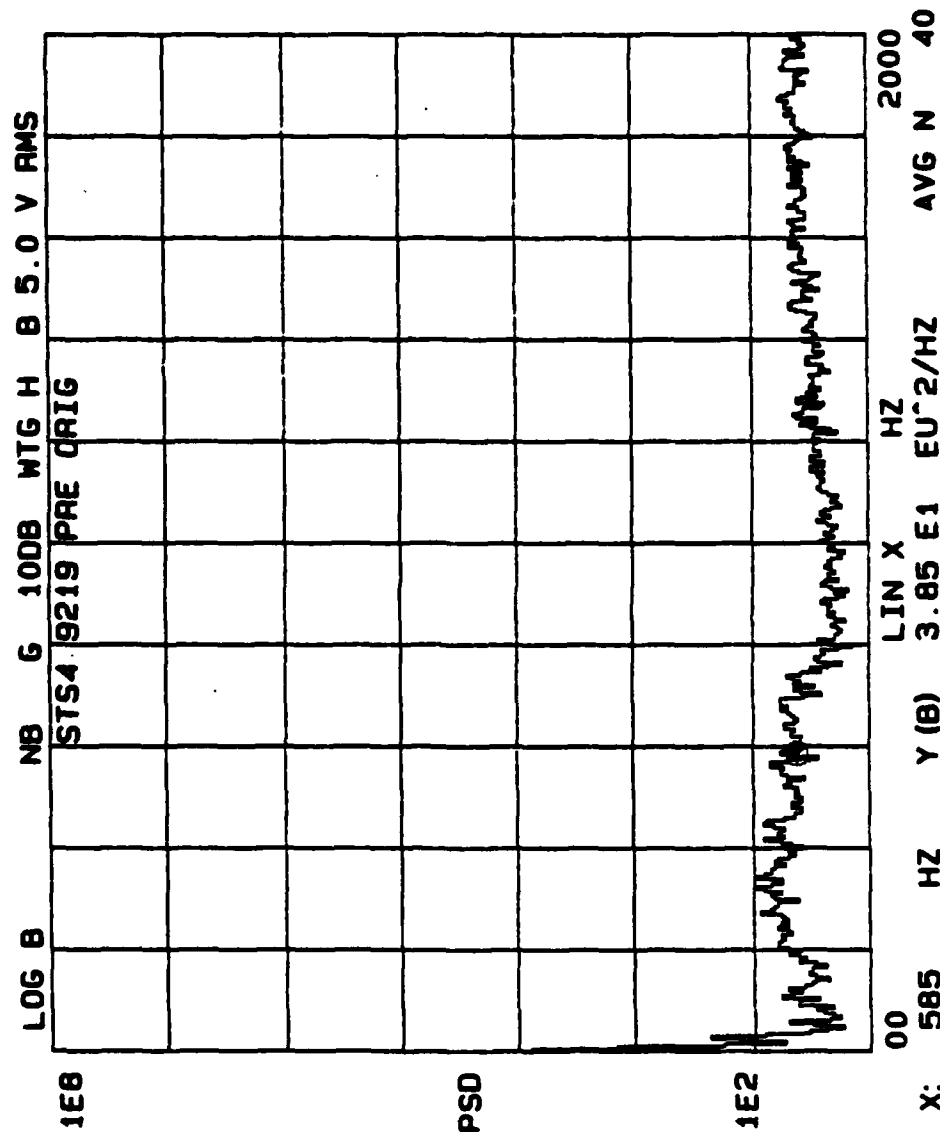
C*****ABP00010
C                                          *ABP00020
C                                          *ABP00030
C                      APPENDIX C          *ABP00040
C                                          *ABP00050
C                      FORTRAN PROGRAM ABPDBP AND PROGRAM OUTPUT *ABP00060
C                                          *ABP00070
C                                          *ABP00080
C          THIS PROGRAM CALCULATES:        *ABP00090
C                                          *ABP00100
C          1) ANALOG ELLIPTIC BANDPASS FILTER RESISTOR AND CAPACITOR *ABP00110
C              VALUES (STARTING FROM TABULATED PARAMETERS CORRESPONDING *ABP00120
C              TO THE DESIRED FILTER RESPONSE) *ABP00130
C                                          *ABP00140
C          2) ANALOG ELLIPTIC BANDPASS TRANSFER FUNCTION (IF DESIRED FOR *ABP00150
C              DIGITAL TRANSFORMATION THEN MUST MODIFY CODE IN THIS PROGRAM *ABP00160
C              SECTION AND SUBSTITUTE WODIG FOR W0 TO REALIZE NECESSARY *ABP00170
C              PRE-WARPING COMPENSATION) *ABP00180
C                                          *ABP00190
C              A) ANALOG TRANSFER FUNCTION COMPLEX ZEROS AND POLES *ABP00200
C                                          *ABP00210
C          3) DIGITAL TRANSFER FUNCTION (BY APPLICATION OF THE BILINEAR *ABP00220
C              TRANSFORM TO THE PRE-WARPED CASE OF THE ANALOG TRANSFER *ABP00230
C              FUNCTION, WHICH IS PROVIDED IN SECTION 2 BY USING THE *ABP00240
C              PRE-WARPED FREQUENCY ANALOG -- WODIG (VICE W0) -- IN THE *ABP00250
C              ANALOG TRANSFER FUNCTION COMPUTATION). THE BILINEAR *ABP00260
C              TRANSFORM IS ACCOMPLISHED BY THE FOLLOWING SUBSTITUTION: *ABP00270
C                                          *ABP00280
C              
$$S = (2/T) \frac{Z - 1}{Z + 1}$$
 *ABP00290
C                                          *ABP00300
C              WHERE T IS THE SAMPLING FREQUENCY OF THE DIGITAL SYSTEM. *ABP00310
C                                          *ABP00320
C              A) DIGITAL TRANSFER FUNCTION COMPLEX ZEROS AND POLES *ABP00330
C                                          *ABP00340
C              *ABP00350
C          4) POLYNOMIAL COEFFICIENTS FOR FIRST AND SECOND ORDER CASCADED *ABP00360
C              TERMS WHICH WILL BE USED TO PERFORM A 2920 ANALOG/DIGITAL *ABP00370
C              SIGNAL PROCESSING SIMULATION *ABP00380
C                                          *ABP00390
C*****ABP00400
C          TYPE DECLARATIONS *ABP00410
C*****ABP00420
C          SECTION 1 *ABP00430
C              REAL A,B,C,D,E,A1,Q,K,K1,K2,W0,PI,FO,WODIG,FODIG,T,TDIV2 *ABP00440
C              REAL C11,C12,C21,C22 *ABP00450
C              REAL R11,R12,R13,R14,R15,R16,R17 *ABP00460
C              REAL R21,R22,R23,R24,R25,R26,R27 *ABP00470
C          SECTION 2 *ABP00480
C              INTEGER INUM1,INUM2,IDEN1,IDEN2,IANUM,IADEN *ABP00490
C              REAL RHO,RHO1,RHO2,ALFA1,ALFA2,BETA1,BETA2,GAMMA1,GAMMA2 *ABP00500
C              REAL NUM1(3),NUM2(3),DEN1(3),DEN2(3) *ABP00510
C              REAL ANUM(5),ADEN(5) *ABP00520
C          SECTION 2A *ABP00530
C              INTEGER IERR,IPDEG *ABP00540
C              REAL ANMINV(5),ADNINV(5) *ABP00550

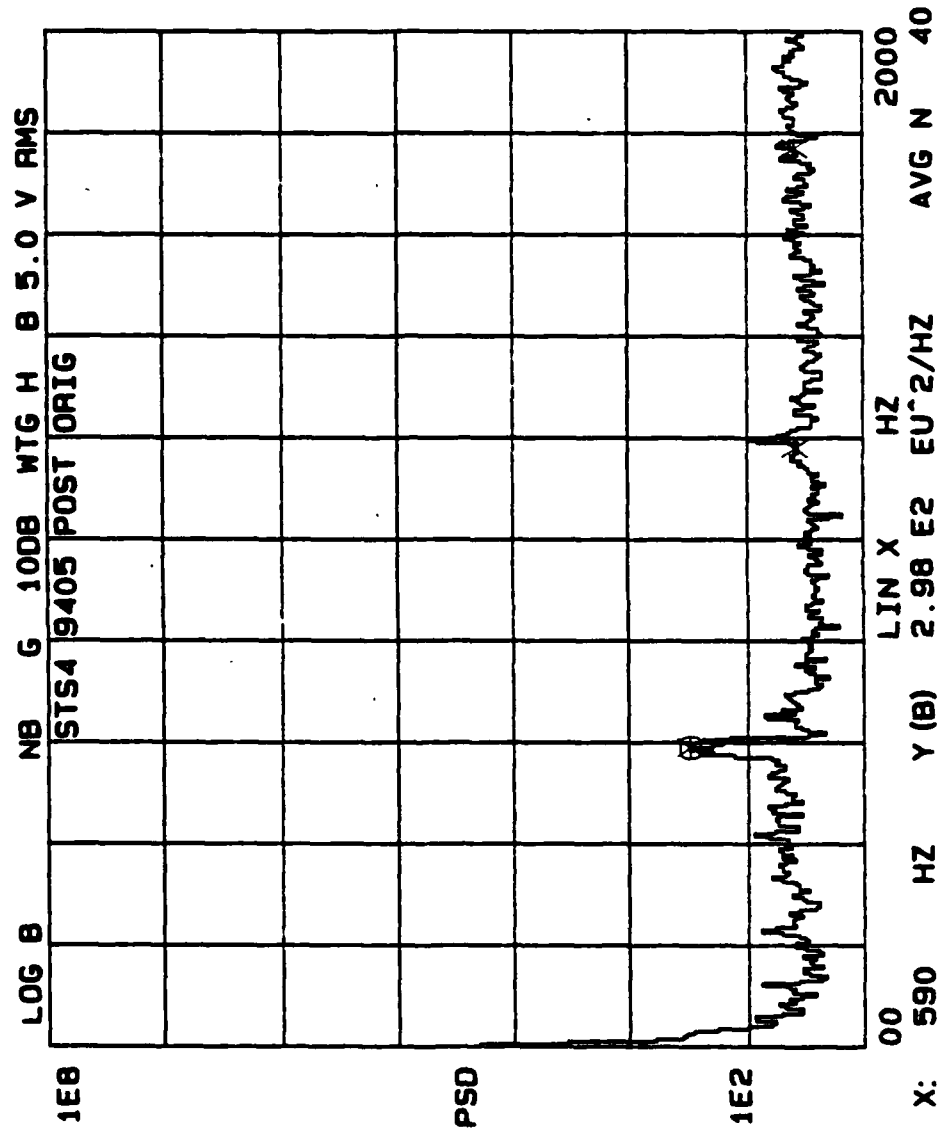
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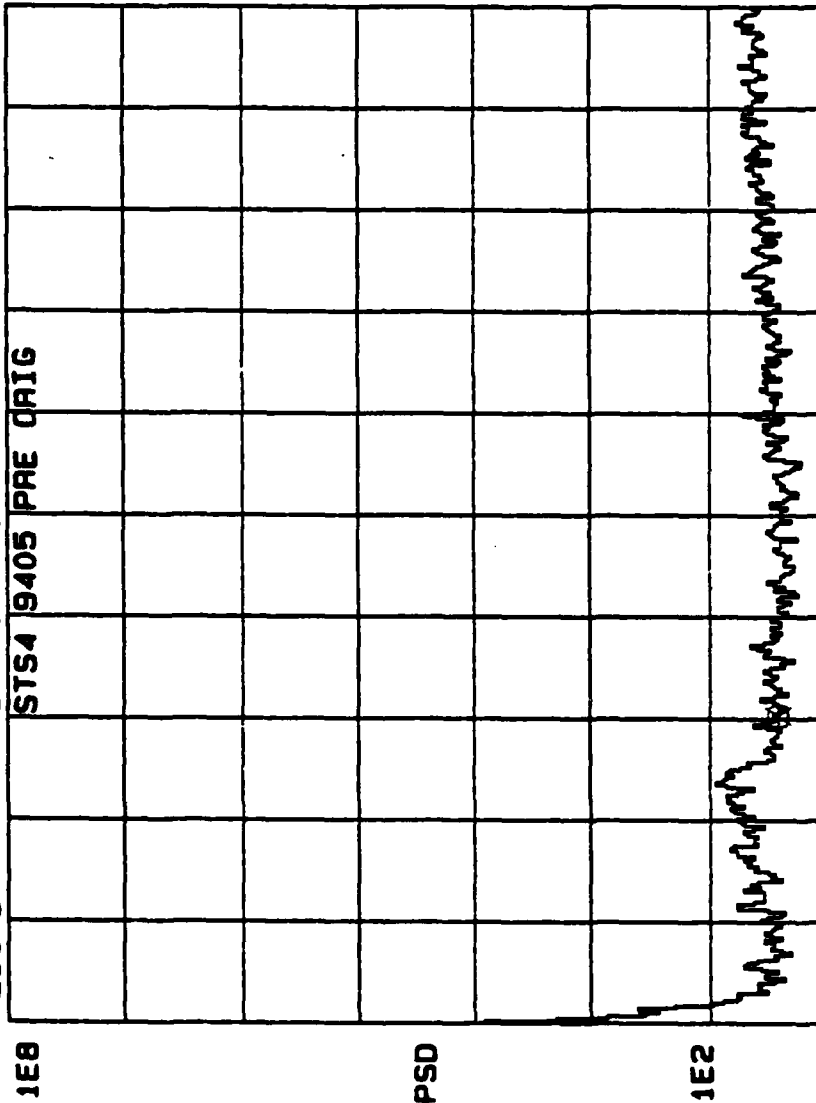




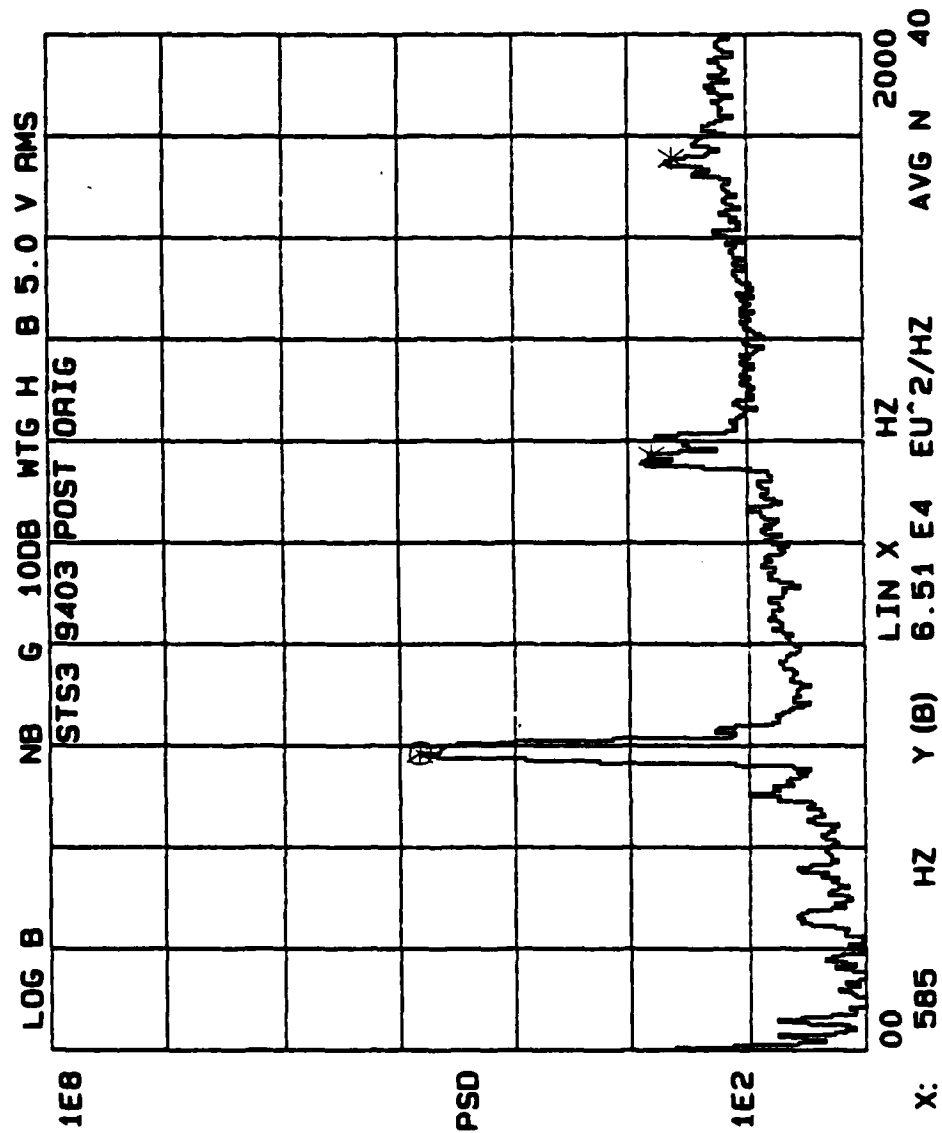


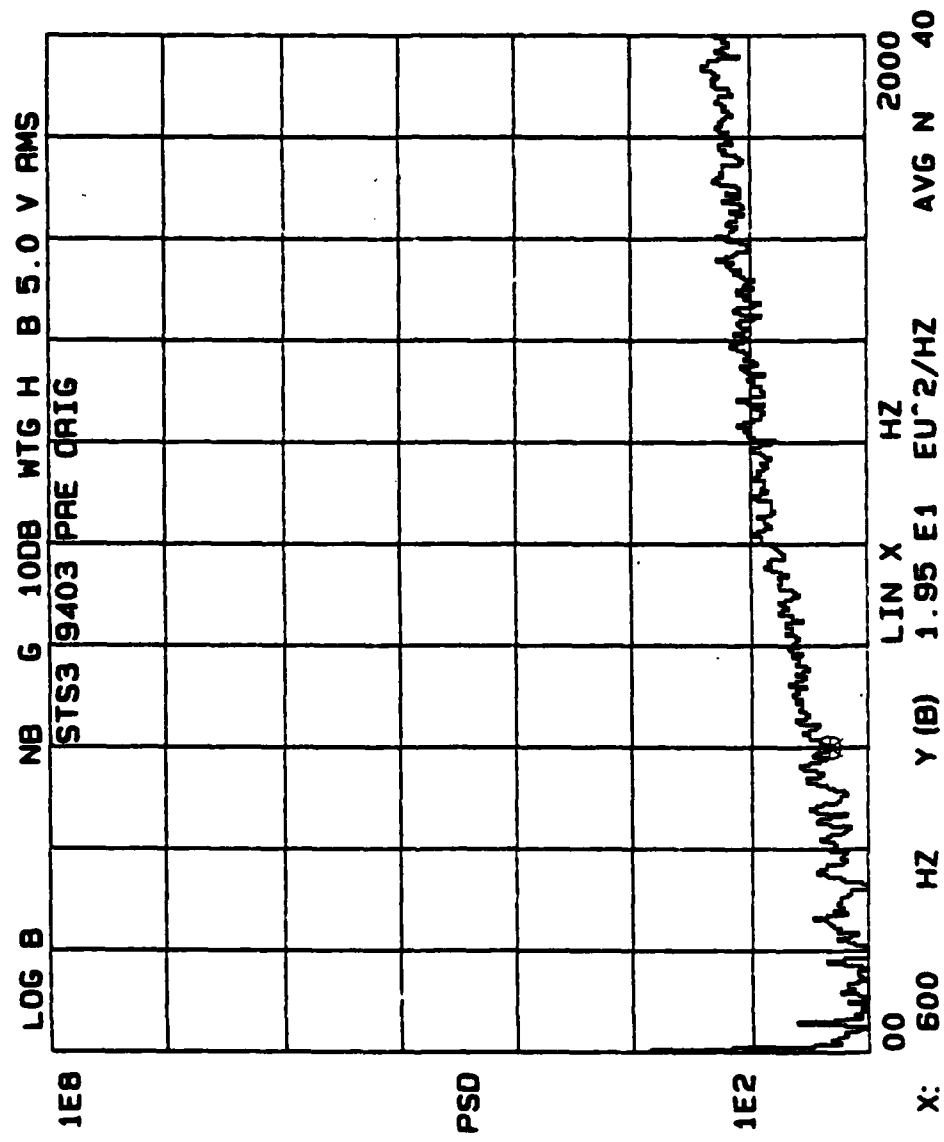


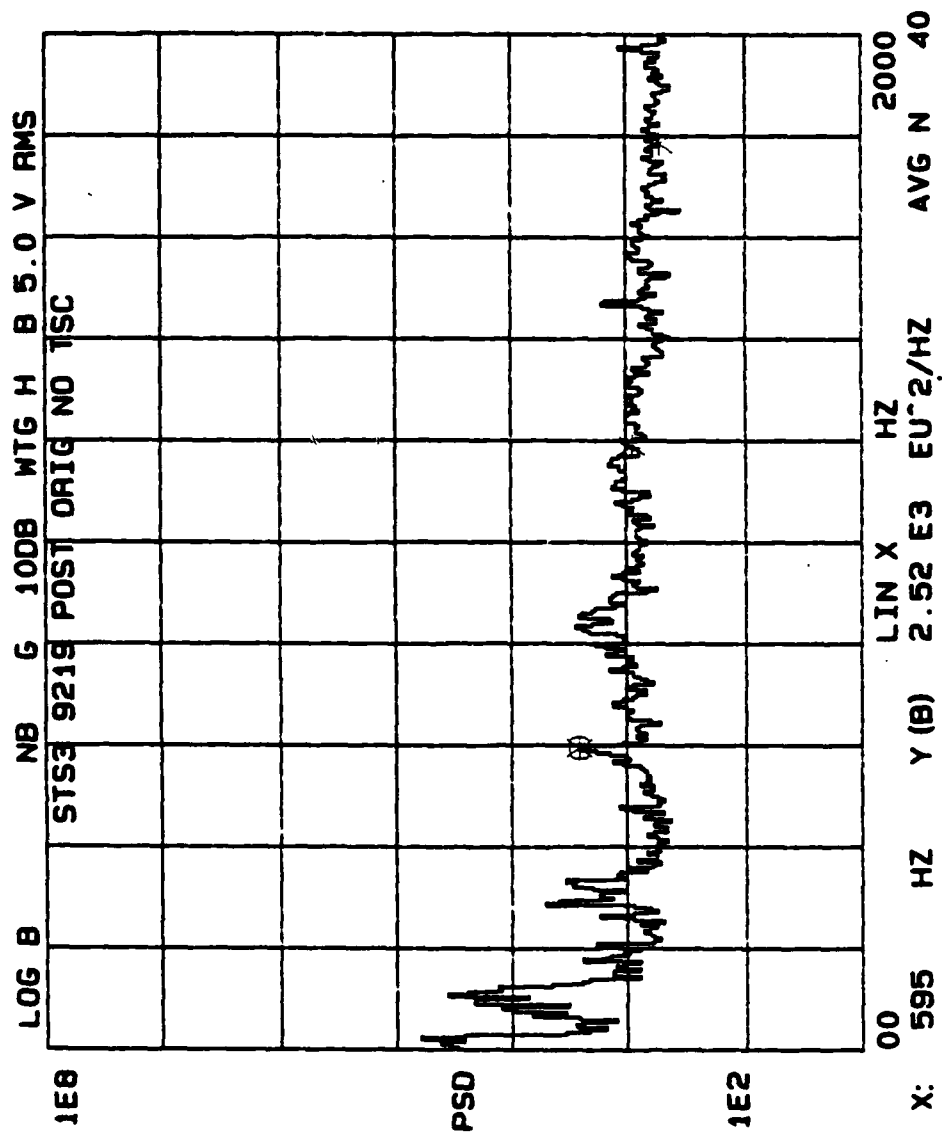
LOG B NB G 100B WTG H B 5.0 V RMS

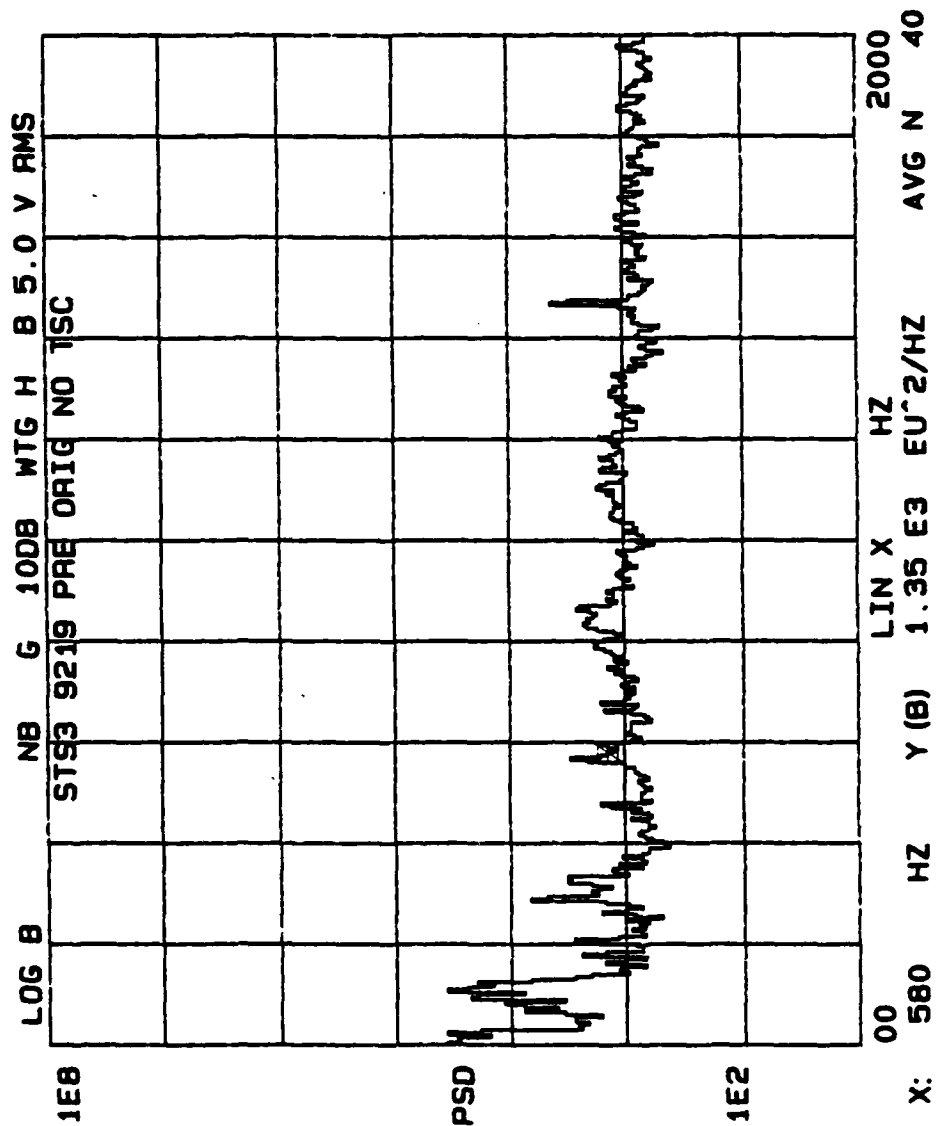


00 605 2000
X: 605 HZ Y (B) 2.48 E1 EU²/HZ AVG N 40









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DO 1400 I=1,5
  ANMINV(I)=ANUM(6-I)
  ADNINV(I)=ADEN(6-I)
1400 CONTINUE
  CALL ZRPOLY(ANMINV,IPDEG,AZERO,IERR)
  CALL ZRPOLY(ADNINV,IPDEG,APOLE,IERR)
  WRITE(4,1410)
  WRITE(4,1420)AZERO(1),AZERO(2),AZERO(3),AZERO(4)
  WRITE(4,1430)APOLE(1),APOLE(2),APOLE(3),APOLE(4)
1410 FORMAT(///,' SECTION 2A OUTPUT',//,
  &' ANALOG FILTER COMPLEX POLES AND ZEROS',/)
1420 FORMAT(' ANALOG FILTER ZEROS',/,4(1X,E12.5,2X,E12.5,/,/),/)
1430 FORMAT(' ANALOG FILTER POLES',/,4(1X,E12.5,2X,E12.5,/,/),/))
C*****
C
C              SECTION 3
C
C  THE FOLLOWING STATEMENTS PERFORM THE BILINEAR TRANSFORMATION
C  OF THE ANALOG BANDPASS TRANSFER FUNCTION COMPUTED ABOVE IN
C  SECTION 2
C*****
C
C  IZM=5
C  IZP=5
C  IDNUM=5
C  IDDEN=5
C  ZM1(1)=-1.
C  ZM1(2)=1.
C  ZM1(3)=0.
C  ZM1(4)=0.
C  ZM1(5)=0.
C  ZM2(1)=1.
C  ZM2(2)=-2.
C  ZM2(3)=1.
C  ZM2(4)=0.
C  ZM2(5)=0.
C  ZM3(1)=-1.
C  ZM3(2)=3.
C  ZM3(3)=-3.
C  ZM3(4)=1.
C  ZM3(5)=0.
C  ZM4(1)=1.
C  ZM4(2)=-4.
C  ZM4(3)=6.
C  ZM4(4)=-4.
C  ZM4(5)=1.
C  ZP1(1)=1.
C  ZP1(2)=1.
C  ZP1(3)=0.
C  ZP1(4)=0.
C  ZP1(5)=0.
C  ZP2(1)=1.
C  ZP2(2)=2.
C  ZP2(3)=1.
C  ZP2(4)=0.

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ZP2(5)=0.	ABP02760
ZP3(1)=1.	ABP02770
ZP3(2)=3.	ABP02780
ZP3(3)=3.	ABP02790
ZP3(4)=1.	ABP02800
ZP3(5)=0.	ABP02810
ZP4(1)=1.	ABP02820
ZP4(2)=4.	ABP02830
ZP4(3)=6.	ABP02840
ZP4(4)=4.	ABP02850
ZP4(5)=1.	ABP02860
TNCOEF=ANUM(1)*(TDIV2**4.)	ABP02870
TDCOEF=ADEN(1)*(TDIV2**4.)	ABP02880
DO 200 I=1,5	ABP02890
DNUM(I)=ZP4(I)*TNCOEF	ABP02900
DDEN(I)=ZP4(I)*TDCOEF	ABP02910
200 CONTINUE	ABP02920
TNCOEF=ANUM(2)*(TDIV2**3.)	ABP02930
TDCOEF=ADEN(2)*(TDIV2**3.)	ABP02940
DO 210 I=1,5	ABP02950
CALL PMPY(DTMP, IDTMP, ZM1, IZM, ZP3, IZP)	ABP02960
DNUM(I)=DNUM(I)+(DTMP(I)*TNCOEF)	ABP02970
DDEN(I)=DDEN(I)+(DTMP(I)*TDCOEF)	ABP02980
210 CONTINUE	ABP02990
TNCOEF=ANUM(3)*(TDIV2**2.)	ABP03000
TDCOEF=ADEN(3)*(TDIV2**2.)	ABP03010
DO 220 I=1,5	ABP03020
CALL PMPY(DTMP, IDTMP, ZM2, IZM, ZP2, IZP)	ABP03030
DNUM(I)=DNUM(I)+(DTMP(I)*TNCOEF)	ABP03040
DDEN(I)=DDEN(I)+(DTMP(I)*TDCOEF)	ABP03050
220 CONTINUE	ABP03060
TNCOEF=ANUM(4)*(TDIV2)	ABP03070
TDCOEF=ADEN(4)*(TDIV2)	ABP03080
DO 230 I=1,5	ABP03090
CALL PMPY(DTMP, IDTMP, ZM3, IZM, ZP1, IZP)	ABP03100
DNUM(I)=DNUM(I)+(DTMP(I)*TNCOEF)	ABP03110
DDEN(I)=DDEN(I)+(DTMP(I)*TDCOEF)	ABP03120
230 CONTINUE	ABP03130
TNCOEF=ANUM(5)	ABP03140
TDCOEF=ADEN(5)	ABP03150
DO 240 I=1,5	ABP03160
DNUM(I)=DNUM(I)+(ZM4(I)*TNCOEF)	ABP03170
DDEN(I)=DDEN(I)+(ZM4(I)*TDCOEF)	ABP03180
240 CONTINUE	ABP03190
DO 250 I=1,5	ABP03200
DNUM(I)=DNUM(I)/DDEN(5)	ABP03210
DDEN(I)=DDEN(I)/DDEN(5)	ABP03220
250 CONTINUE	ABP03230
RHO=RHO*DNUM(5)	ABP03240
DO 260 I=1,5	ABP03250
DNUM(I)=DNUM(I)/DNUM(5)	ABP03260
260 CONTINUE	ABP03270
WRITE(4,310)RHO	ABP03280
WRITE(4,320)DNUM(5),DNUM(4),DNUM(3),DNUM(2),DNUM(1)	ABP03290
DO 305 I=1,5	ABP03300


```

14=4
12=2
DO 500 I=1,4
  DZ4(I)=CMPLX(REAL(DZERO(I)),AIMAG(DZERO(I)))
  DP4(I)=CMPLX(REAL(DPOLE(I)),AIMAG(DPOLE(I)))
500 CONTINUE
  DO 502 I=1,2
    DZ21(I)=DZ4(I)
    DP21(I)=DP4(I)
502 CONTINUE
  DO 504 I=3,4,1
    DZ22(I-2)=DZ4(I)
    DP22(I-2)=DP4(I)
504 CONTINUE
  CALL MAKPOL(14,DZ4,DP4)
  CALL MAKPOL(14,DP4,DD4)
  DN45(5)=CMPLX(1.,0.)
  DD45(5)=CMPLX(1.,0.)
  DO 512 I=1,4
    DN45(I)=DN4(I)
    DD45(I)=DD4(I)
512 CONTINUE
  WRITE(4,514)
514 FORMAT('1', 'SECTION 4 OUTPUT', //,
&' REASSEMBLE COEFFICIENTS FROM POLES AND ZEROS', //)
  WRITE(4,515)
  WRITE(4,520)DN45(5),DN45(4),DN45(3),DN45(2),DN45(1)
  WRITE(4,530)DD45(5),DD45(4),DD45(3),DD45(2),DD45(1)
515 FORMAT(/' SINGLE FOURTH ORDER TRANSFER FUNCTION COEFFICIENTS', //)
520 FORMAT(' FOURTH ORDER NUMERATOR COEFFICIENTS (NORMALIZED)', //,
&5(1X,E12.5,2X,E12.5, //))
530 FORMAT(' FOURTH ORDER DENOMINATOR COEFFICIENTS (NORMALIZED)', //,
&5(1X,E12.5,2X,E12.5, //, //))
  CALL MAKPOL(12,DZ21,DP21)
  CALL MAKPOL(12,DP21,DD21)
  DN213(3)=CMPLX(1.,0.)
  DD213(3)=CMPLX(1.,0.)
  DO 612 I=1,2
    DN213(I)=DN21(I)
    DD213(I)=DD21(I)
612 CONTINUE
  WRITE(4,614)
614 FORMAT(' CASCADED SECOND ORDER TRANSFER FUNCTION COEFFICIENTS', //)
  WRITE(4,615)
  WRITE(4,620)DN213(3),DN213(2),DN213(1)
  WRITE(4,630)DD213(3),DD213(2),DD213(1)
615 FORMAT(/' FIRST STAGE QUADRATIC FUNCTION COEFFICIENTS', //)
620 FORMAT(' COMPLEX (REAL,IMAG) NUMERATOR COEFFICIENTS', //,
&1X,E12.5,2X,E12.5,
&' Z**2 +', //,1X,E12.5,2X,E12.5, ' Z +', //,1X,E12.5,2X,E12.5, //)
630 FORMAT(' COMPLEX (REAL,IMAG) DENOMINATOR COEFFICIENTS', //,
&1X,E12.5,2X,E12.5,
&' Z**2 +', //,1X,E12.5,2X,E12.5, ' Z +', //,1X,E12.5,2X,E12.5, //)
  CALL MAKPOL(12,DZ22,DP22)
  CALL MAKPOL(12,DP22,DD22)

```

ABP03860
ABP03870
ABP03880
ABP03890
ABP03900
ABP03910
ABP03920
ABP03930
ABP03940
ABP03950
ABP03960
ABP03970
ABP03980
ABP03990
ABP04000
ABP04010
ABP04020
ABP04030
ABP04040
ABP04050
ABP04060
ABP04070
ABP04080
ABP04090
ABP04100
ABP04110
ABP04120
ABP04130
ABP04140
ABP04150
ABP04160
ABP04170
ABP04180
ABP04190
ABP04200
ABP04210
ABP04220
ABP04230
ABP04240
ABP04250
ABP04260
ABP04270
ABP04280
ABP04290
ABP04300
ABP04310
ABP04320
ABP04330
ABP04340
ABP04350
ABP04360
ABP04370
ABP04380
ABP04390
ABP04400

```

DN223(3)=CMPLX(1.,0.)
DD223(3)=CMPLX(1.,0.)
DO 712 I=1,2
    DN223(I)=DN22(I)
    DD223(I)=DD22(I)
712 CONTINUE
WRITE(4,715)
WRITE(4,720)DN223(3),DN223(2),DN223(1)
WRITE(4,730)DD223(3),DD223(2),DD223(1)
715 FORMAT(/' SECOND STAGE QUADRATIC FUNCTION COEFFICIENTS',/)
720 FORMAT(' COMPLEX (REAL,IMAG) NUMERATOR COEFFICIENTS',/,
&1X,E12.5,2X,E12.5,
&' Z**2 +',/,1X,E12.5,2X,E12.5,' Z +',/,1X,E12.5,2X,E12.5,/)
730 FORMAT(' COMPLEX (REAL,IMAG) DENOMINATOR COEFFICIENTS',/,
&1X,E12.5,2X,E12.5,
&' Z**2 +',/,1X,E12.5,2X,E12.5,' Z +',/,1X,E12.5,2X,E12.5,/)
STOP
END
C
C
C .....
C
C SUBROUTINE PMPY
C
C PURPOSE
C     MULTIPLY TWO POLYNOMIALS
C
C USAGE
C     CALL PMPY(Z, IDIMZ, X, IDIMX, Y, IDIMY)
C
C DESCRIPTION OF PARAMETERS
C     Z - VECTOR OF RESULTANT COEFFICIENTS, ORDERED FROM
C         SMALLEST TO LARGEST POWER
C     IDIMZ - DIMENSION OF Z (CALCULATED)
C     X - VECTOR OF COEFFICIENTS FOR FIRST POLYNOMIAL, ORDERED
C         FROM SMALLEST TO LARGEST POWER
C     IDIMX - DIMENSION OF X (DEGREE IS IDIMX-1)
C     Y - VECTOR OF COEFFICIENTS FOR SECOND POLYNOMIAL,
C         ORDERED FROM SMALLEST TO LARGEST POWER
C     IDIMY - DIMENSION OF Y (DEGREE IS IDIMY-1)
C
C REMARKS
C     Z CANNOT BE IN THE SAME LOCATION AS X
C     Z CANNOT BE IN THE SAME LOCATION AS Y
C
C SUBROUTINES AND FUNCTION SUBPROGRAMS REQUIRED
C     NONE
C
C METHOD
C     DIMENSION OF Z IS CALCULATED AS IDIMX+IDIMY-1
C     THE COEFFICIENTS OF Z ARE CALCULATED AS SUM OF PRODUCTS
C     OF COEFFICIENTS OF X AND Y, WHOSE EXPONENTS ADD UP TO THE
C     CORRESPONDING EXPONENT OF Z.
C
C .....
C

```

ABP04410
ABP04420
ABP04430
ABP04440
ABP04450
ABP04460
ABP04470
ABP04480
ABP04490
ABP04500
ABP04510
ABP04520
ABP04530
ABP04540
ABP04550
ABP04560
ABP04570
ABP04580
ABP04590
ABP04600
ABP04610
ABP04620
ABP04630
ABP04640
ABP04650
ABP04660
ABP04670
ABP04680
ABP04690
ABP04700
ABP04710
ABP04720
ABP04730
ABP04740
ABP04750
ABP04760
ABP04770
ABP04780
ABP04790
ABP04800
ABP04810
ABP04820
ABP04830
ABP04840
ABP04850
ABP04860
ABP04870
ABP04880
ABP04890
ABP04900
ABP04910
ABP04920
ABP04930
ABP04940
ABP04950

	SUBROUTINE PMPY(Z, IDIMZ, X, IDIMX, Y, IDIMY)	ABP04960
	DIMENSION Z(1), X(1), Y(1)	ABP04970
C	IF (IDIMX*IDIMY) 10, 10, 20	ABP04980
	10 IDIMZ=0	ABP04990
	GO TO 50	ABP05000
	20 IDIMZ=IDIMX+IDIMY-1	ABP05010
	DO 30 I=1, IDIMZ	ABP05020
	30 Z(I)=0.	ABP05030
	DO 40 I=1, IDIMX	ABP05040
	DO 40 J=1, IDIMY	ABP05050
	K=I+J-1	ABP05060
	40 Z(K)=X(I)*Y(J)+Z(K)	ABP05070
	50 RETURN	ABP05080
	END	ABP05090
C		ABP05100
C		ABP05110
C	-----	ABP05120
C		ABP05130
C	SUBROUTINE MAKPOL	ABP05140
C	PURPOSE	ABP05150
C	TO COMPUTE THE COMPLEX COEFFICIENTS OF AN N-TH DEGREE POLY-	ABP05160
C	NOMIAL GIVEN N COMPLEX ROOTS OF THE POLYNOMIAL	ABP05170
C		ABP05180
C	USAGE	ABP05190
C	CALL MAKPOL(N, R, C)	ABP05200
C		ABP05210
C	DESCRIPTION OF PARAMETERS	ABP05220
C	N - NUMBER OF ROOTS GIVEN AND DEGREE OF POLYNOMIAL. THE	ABP05230
C	COEFFICIENT OF THE HIGHEST POWER OF THE UNKNOWN IS ALWAYS	ABP05240
C	UNITY, AND IS NOT COMPUTED BY "MAKPOL".	ABP05250
C	R - DOUBLE PRECISION COMPLEX ARRAY CONTAINING THE COMPLEX	ABP05260
C	ROOTS	ABP05270
C	C - DOUBLE PRECISION COMPLEX ARRAY CONTAINING THE COMPLEX	ABP05280
C	COEFFICIENTS	ABP05290
C		ABP05300
C	REMARKS	ABP05310
C	ARRAYS R AND C ARE TYPED COMPLEX*16	ABP05320
C		ABP05330
C		ABP05340
C	SUBROUTINE MAKPOL(N, R, C)	ABP05350
	COMPLEX*16 R(N), C(N)	ABP05360
	IF (N.LE.0) RETURN	ABP05370
	DO 10 I=1, N	ABP05380
10	C(I)=R(I)	ABP05390
	K=N	ABP05400
	M=N-1	ABP05410
	DO 20 L=1, M	ABP05420
	DO 30 I=2, K	ABP05430
30	C(I)=C(I)+C(I-1)	ABP05440
	K=K-1	ABP05450
	DO 20 I=1, K	ABP05460
	J=I+L	ABP05470
	C(I)=R(J)*C(I)	ABP05480
20	CONTINUE	ABP05490
	K=N/2	ABP05500

FILE: ABPD8P FORTRAN A1

```
K=2*K/(N-K)
DO 40 I=K,N,2
40 C(I)=-C(I)
RETURN
END
```

ABP05510
ABP05520
ABP05530
ABP05540
ABP05550

ABPOBP OUTPUT

SECTION 1 OUTPUT

INPUT AND DERIVED PARAMETERS FOR FURTHER CALCULATIONS

A = 21.164001
B = 0.787152
C = 0.842554
D = 1.035090
E = 30.507828
A1 = 1.463835
F0 = 600.000
W0 = 3769.911
Q = 12.000000
K = 1.000000
K1 = 1.000000
K2 = 1.000000
W0(DIG) = 3712.266
F0(DIG) = 590.825

ELLIPTIC ANALOG BPF COMPONENT VALUES

FIRST STAGE

C11 = .996E-08
C12 = .995E-08
R17 = .267E+05
R11 = .393E+07
R12 = .785E+06
R13 = .257E+05
R14 = .134E+06
R15 = .945E+05
R16 = .258E+05

SECOND STAGE

C21 = .103E-07
C22 = .103E-07
R27 = .267E+05
R21 = .408E+07
R22 = .813E+06
R23 = .267E+05
R24 = .134E+06
R25 = .182E+06
R26 = .266E+05

SECTION 2 OUTPUT

ANALOG ELLIPTIC BANDPASS FILTER TRANSFER FUNCTION

NUMERATOR COEFFICIENT = RHO = 0.039811

NUMERATOR POLYNOMIAL (NORMALIZED)

0.10000E+01 S**4 +
0.0 S**3 +
0.29587E+08 S**2 +
0.0 S +
0.18991E+15

NUMERATOR POLYNOMIAL (UN-NORMALIZED)

0.39811E-01 S**4 +
0.0 S**3 +
0.11779E+07 S**2 +
0.0 S +
0.75606E+13

DENOMINATOR POLYNOMIAL (NORMALIZED)

0.10000E+01 S**4 +
0.24351E+03 S**3 +
0.27642E+08 S**2 +
0.33558E+10 S +
0.18991E+15

SECTION 2A OUTPUT

ANALOG FILTER COMPLEX POLES AND ZEROS

ANALOG FILTER ZEROS

-0.45776E-03 0.30683E+04
-0.45776E-03 -0.30683E+04
0.45776E-03 0.44914E+04
0.45776E-03 -0.44914E+04

ANALOG FILTER POLES

-0.58188E+02 0.35859E+04
-0.58188E+02 -0.35859E+04
-0.63567E+02 0.38421E+04
-0.63567E+02 -0.38421E+04

SECTION 3 OUTPUT

EQUIVALENT DIGITAL FILTER TRANSFER FUNCTION

NUMERATOR COEFFICIENT = 0.39516E-01

NUMERATOR (NORMALIZED)

0.10000E+01 +
-0.36279E+01 Z**-1 +
0.52861E+01 Z**-2 +
-0.36279E+01 Z**-3 +
0.10000E+01 Z**-4

NUMERATOR (UN-NORMALIZED)

0.39516E-01 +
-0.14336E+00 Z**-1 +
0.20888E+00 Z**-2 +
-0.14336E+00 Z**-3 +
0.39516E-01 Z**-4

DENOMINATOR

0.10000E+01 +
-0.36251E+01 Z**-1 +
0.52586E+01 Z**-2 +
-0.35768E+01 Z**-3 +
0.97353E+00 Z**-4

SECTION 3A OUTPUT

DIGITAL FILTER.COMPLEX POLES AND ZEROS AND RADIUS

ZERO LOCATIONS (REAL,IMAG) AND RADIUS
0.87496E+00 0.48595E+00 0.10009E+01
0.87496E+00 -0.48595E+00 0.10009E+01
0.93896E+00 0.34135E+00 0.99909E+00
0.93896E+00 -0.34135E+00 0.99909E+00

POLE LOCATIONS (REAL,IMAG) AND RADIUS
0.90781E+00 0.40813E+00 0.99533E+00
0.90781E+00 -0.40813E+00 0.99533E+00
0.90475E+00 0.40493E+00 0.99123E+00
0.90475E+00 -0.40493E+00 0.99123E+00

SECTION 4 OUTPUT

REASSEMBLE COEFFICIENTS FROM POLES AND ZEROS

SINGLE FOURTH ORDER TRANSFER FUNCTION COEFFICIENTS

FOURTH ORDER NUMERATOR COEFFICIENTS (NORMALIZED)

0.10000E+01	0.0
-0.36279E+01	0.0
0.52861E+01	0.0
-0.36279E+01	0.0
0.99988E+00	0.0

FOURTH ORDER DENOMINATOR COEFFICIENTS (NORMALIZED)

0.10000E+01	0.0
-0.36251E+01	0.0
0.52586E+01	0.0
-0.35766E+01	-0.18041E-15
0.97339E+00	0.0

CASCADED SECOND ORDER TRANSFER FUNCTION COEFFICIENTS

FIRST STAGE QUADRATIC FUNCTION COEFFICIENTS

COMPLEX (REAL, IMAG) NUMERATOR COEFFICIENTS

0.10000E+01	0.0	Z**2 +
-0.17499E+01	0.0	Z +
0.10017E+01	0.0	

COMPLEX (REAL, IMAG) DENOMINATOR COEFFICIENTS

0.10000E+01	0.0	Z**2 +
-0.18156E+01	0.0	Z +
0.99068E+00	0.0	

SECOND STAGE QUADRATIC FUNCTION COEFFICIENTS

COMPLEX (REAL, IMAG) NUMERATOR COEFFICIENTS

0.10000E+01	0.0	Z**2 +
-0.18779E+01	0.0	Z +
0.99817E+00	0.0	

COMPLEX (REAL, IMAG) DENOMINATOR COEFFICIENTS

0.10000E+01	0.0	Z**2 +
-0.18095E+01	0.0	Z +
0.98255E+00	0.0	

```

C*****ABP00010
C                                          *ABP00020
C                      APPENDIX D          *ABP00030
C                                          *ABP00040
C                      FORTRAN PROGRAM ABPFR *ABP00050
C                                          *ABP00060
C                                          *ABP00070
C PROGRAM TO PLOT ANALOG BAND-PASS FILTER FREQUENCY AND PHASE *ABP00080
C RESPONSE OF THE ELLIPTIC FILTER TRANSFER FUNCTION *ABP00090
C                                          *ABP00100
C*****ABP00110
C TYPE DECLARATIONS                      ABP00120
C   IMPLICIT REAL(A-H,O-Z), INTEGER(I-N) ABP00130
C   REAL W(1001), HMAG(1001), HMAGN(1001), HMAGDB(1001), WX, F(1001) ABP00140
C   REAL A(5), B(5), HPHASE(1001)        ABP00150
C   COMPLEX S, H                          ABP00160
C NORMALIZED ANALOG TRANSFER FUNCTION COEFFICIENTS ABP00170
C   A(1) = 1.                             ABP00180
C   A(2) = 0.0                             ABP00190
C   A(3) = 0.30513E+8                      ABP00200
C   A(4) = 0.0                             ABP00210
C   A(5) = 0.20198E+15                     ABP00220
C   B(1) = 1.0                             ABP00230
C   B(2) = 0.24729E+3                      ABP00240
C   B(3) = 0.28507E+8                      ABP00250
C   B(4) = 0.35145E+10                     ABP00260
C   B(5) = 0.20198E+15                     ABP00270
C CONSTANTS                              ABP00280
C   PI = 3.1415927                         ABP00290
C EVALUATE MAGNITUDE AND PHASE OF H(JW) ABP00300
C   DO 10 I = 1, 1001                      ABP00310
C     F(I) = FLOAT(I-1)                    ABP00320
C     W(I) = (2.*PI*F(I))                  ABP00330
C     S = CMPLX(0., W(I))                  ABP00340
C     H = (A(1)*(S**4)+A(2)*(S**3)+A(3)*(S**2)+A(4)*S+A(5))/ ABP00350
C     &(B(1)*(S**4)+B(2)*(S**3)+B(3)*(S**2)+B(4)*S+B(5)) ABP00360
C     HMAG(I) = CABS(H)                    ABP00370
C     X = REAL(H)                          ABP00380
C     Y = AIMAG(H)                         ABP00390
C     HPHASE(I) = ATAN(Y/X)*180./PI        ABP00400
C 10 CONTINUE                             ABP00410
C NORMALIZE MAGNITUDE                     ABP00420
C   AMAX = 0.0                             ABP00430
C   DO 20 I = 1, 1001                      ABP00440
C     IF(HMAG(I).GT.AMAX) AMAX = HMAG(I) ABP00450
C 20 CONTINUE                             ABP00460
C   DO 30 I = 1, 1001                      ABP00470
C     HMAGN(I) = HMAG(I)/AMAX              ABP00480
C     HMAGDB(I) = 20.0 * ALOG10(HMAG(I)) ABP00490
C 30 CONTINUE                             ABP00500
C   DO 40 I = 1, 1001                      ABP00510
C     WRITE (4,50) I, F(I), W(I), HMAG(I), HMAGN(I), HMAGDB(I), HPHASE(I) ABP00520
C 50 FORMAT(14,6(1X,E10.3))              ABP00530
C 40 CONTINUE                             ABP00540
C-----ABP00550

```

C GRAPHICS PARAMETERS FOR MAGNITUDE VS FREQUENCY	ABP00560
C-----	ABP00570
CALL LRGBUF	ABP00580
CALL COMPRS	ABP00590
C CALL TEK618	ABP00600
C CALL VRSTEC(0,0,0)	ABP00610
C SETUP THE PLOTTING AREA	ABP00620
CALL PAGE (11.0,8.5)	ABP00630
CALL NOBRDR	ABP00640
CALL AREA2D(9.0,6.5)	ABP00650
C LABEL THE X & Y AXES	ABP00660
CALL XNAME('FREQUENCY (HZ)\$',100)	ABP00670
CALL YNAME('AMPLITUDES',100)	ABP00680
CALL HEADIN('ANALOG ELLIPTIC BPF FREQUENCY RESPONSES',100,1.6,2)	ABP00690
CALL HEADIN('AMPLITUDE VS FREQ (F0=600 HZ)\$',100,1.,2)	ABP00700
C DEFINE THE AXES	ABP00710
CALL GRAF(0.0,'SCALE',1000.,-0.5,'SCALE',+1.5)	ABP00720
C DRAW THE CURVES	ABP00730
C CALL THKCRV(0.02)	ABP00740
CALL MARKER(15)	ABP00750
CALL CURVE(F,HMAGN,1001,0)	ABP00760
C TERMINATE THIS PLOT	ABP00770
CALL ENDPL(0)	ABP00780
C-----	ABP00790
C GRAPHICS PARAMETERS FOR MAGNITUDE IN DBS VS FREQUENCY	ABP00800
C-----	ABP00810
CALL LRGBUF	ABP00820
CALL COMPRS	ABP00830
C CALL TEK618	ABP00840
C CALL VRSTEC(0,0,0)	ABP00850
C SETUP THE PLOTTING AREA	ABP00860
CALL PAGE (11.0,8.5)	ABP00870
CALL NOBRDR	ABP00880
CALL AREA2D(9.0,6.5)	ABP00890
C LABEL THE X & Y AXES	ABP00900
CALL XNAME('FREQUENCY (HZ)\$',100)	ABP00910
CALL YNAME('AMPLITUDE (DB)\$',100)	ABP00920
CALL HEADIN('ANALOG ELLIPTIC BPF FREQUENCY RESPONSES',100,1.6,2)	ABP00930
CALL HEADIN('AMPLITUDE (DB) VS FREQ (F0=600 HZ)\$',100,1.,2)	ABP00940
C DEFINE THE AXES	ABP00950
CALL GRAF(0.0,'SCALE',1000.,-60.0,'SCALE',30.0)	ABP00960
C DRAW THE CURVES	ABP00970
C CALL THKCRV(0.02)	ABP00980
CALL MARKER(15)	ABP00990
CALL CURVE(F,HMAGDB,1001,0)	ABP01000
C TERMINATE THIS PLOT	ABP01010
CALL ENDPL(0)	ABP01020
C-----	ABP01030
C GRAPHICS PARAMETERS FOR PHASE VS FREQUENCY	ABP01040
C-----	ABP01050
CALL LRGBUF	ABP01060
CALL COMPRS	ABP01070
C CALL TEK618	ABP01080
C CALL VRSTEC(0,0,0)	ABP01090
C SETUP THE PLOTTING AREA	ABP01100

FILE: ABPFR FORTRAN A1

CALL PAGE (11.0,8.5)	ABP01110
CALL NOBRDR	ABP01120
CALL AREA2D(9.0,6.5)	ABP01130
C LABEL THE X & Y AXES	ABP01140
CALL XNAME('FREQUENCY (HZ)\$',100)	ABP01150
CALL YNAME('PHASE (DEGREES)\$',100)	ABP01160
CALL HEADIN('ANALOG ELLIPTIC BPF PHASE RESPONSES',100,1.6,2)	ABP01170
CALL HEADIN('PHASE VS FREQ (F0=600 HZ)\$',100,1.,2)	ABP01180
C DEFINE THE AXES	ABP01190
CALL GRAF(0.0,'SCALE',1000.,-100.,'SCALE',100.)	ABP01200
C DRAW THE CURVES	ABP01210
C CALL THKCRV(0.02)	ABP01220
CALL MARKER(15)	ABP01230
CALL CURVE(F,HPHASE,1001,0)	ABP01240
C TERMINATE THIS PLOT	ABP01250
CALL ENDPL(0)	ABP01260
CALL DONEPL	ABP01270
STOP	ABP01280
END	ABP01290

```

      DN223(1)=0.998169
      DD223(3)=1.
      DD223(2)=-1.809446
      DD223(1)=0.982544
C PRINT SECOND STAGE TRANSFER FUNCTION
      WRITE(4,82)DN223(3),DN223(2),DN223(1),DD223(3),DD223(2),DD223(1)
      82 FORMAT(' SECOND STAGE TRANSFER FUNCTION (2920 EQUIVALENT)',//,
&' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/,
&' -----',/,
&' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///)
C INITIALIZE VARIABLES
      PI=3.1415927
      SCALS1=0.
      SCALS2=0.
C SAMPLE PERIOD = T = 1.1521152 X 10**-4 SECONDS
      T=4.*192./6.666E6
C COMPUTE SCALE SUM FOR STAGES TO LIMIT INPUT AMPLITUDE
      DO 6 K=1,3
        SCALS1=SCALS1+ABS(DN213(K))
        SCALS1=SCALS1+ABS(DD213(K))
        SCALS2=SCALS2+ABS(DN223(K))
        SCALS2=SCALS2+ABS(DD223(K))
      6 CONTINUE
C ENSURE SCALE SUM FACTORS WILL LIMIT OUTPUT TO LESS THAN ONE
      SCALS1=SCALS1+1.E-6
      SCALS2=SCALS2+1.E-6
C COMPUTE SIMULATED INPUT MAGNITUDE LIMIT
      SINMAG=1./(SCALS1*SCALS2)
C PRINT STAGE SCALE SUMS AND INPUT MAGNITUDE LIMIT
      WRITE(4,85)SCALS1,SCALS2,SINMAG
      85 FORMAT(//,' SCALE FACTORS AND INPUT MAGNITUDE LIMIT',//,
&' FIRST STAGE SCALE SUM = ',E14.6,/,
&' SECOND STAGE SCALE SUM = ',E14.6,/,
&' INPUT AMPLITUDE LIMITED TO +/- ',E14.6,///)
      WRITE(4,86)
      86 FORMAT('1')
C BEGIN SIMULATION FOR SPECIFIED FREQUENCIES GIVEN BY F(L)
C*****
C ADJUST AS NECESSARY
      DO 20 L=1,6
C*****
C COMPUTE SIMULATION RUN INPUT CONSTANT FOR EACH FREQUENCY
      TWOPIF=2.*PI*F(L)
C INITIALIZE STAGE INPUTS AND OUTPUTS
      IMOUT=0
      MOUT=0.
      DO 5 I=1,3
        X1(I)=0.
        X2(I)=0.
        Y1(I)=0.
        Y2(I)=0.
      5 CONTINUE
C PRINT SIMULATION HEADINGS
      WRITE(4,98)F(L)
      98 FORMAT(' FILTER FREQUENCY RESPONSE FOR F = ',F5.0,' HZ',/)

```

S2200560
 S2200570
 S2200580
 S2200590
 S2200600
 S2200610
 S2200620
 S2200630
 S2200640
 S2200650
 S2200660
 S2200670
 S2200680
 S2200690
 S2200700
 S2200710
 S2200720
 S2200730
 S2200740
 S2200750
 S2200760
 S2200770
 S2200780
 S2200790
 S2200800
 S2200810
 S2200820
 S2200830
 S2200840
 S2200850
 S2200860
 S2200870
 S2200880
 S2200890
 S2200900
 S2200910
 S2200920
 S2200930
 S2200940
 S2200950
 S2200960
 S2200970
 S2200980
 S2200990
 S2201000
 S2201010
 S2201020
 S2201030
 S2201040
 S2201050
 S2201060
 S2201070
 S2201080
 S2201090
 S2201100

```

C*****S2200010
C                                     *S2200020
C                                     *S2200030
C                                     *S2200040
C                                     *S2200050
C                                     *S2200060
C                                     *S2200070
C                                     *S2200080
C                                     *S2200090
C                                     *S2200100
C                                     *S2200110
C                                     *S2200120
C                                     *S2200130
C                                     *S2200140
C                                     *S2200150
C                                     *S2200160
C                                     *S2200170
C                                     *S2200180
C*****S2200190
C VARIABLE DECLARATIONS
C   INTEGER IMOUT
C   REAL TX,INO,OUTO,F(9)
C   REAL X1(3),X2(3),Y1(3),Y2(3)
C   REAL SCALS1,SCALS2,SINMAG,MOUT
C   REAL DN213(3),DD213(3),DN223(3),DD223(3),T,PI,TWOPI F
C INPUT FREQUENCIES
C   F(1)=500.
C   F(2)=575.
C   F(3)=590.825
C   F(4)=600.
C   F(5)=625.
C   F(6)=700.
C   F(7)=
C   F(8)=
C   F(9)=
C   WRITE(4,80)
C   80 FORMAT(' PROGRAM S22F OUTPUT',//,
C     &' FOURTH ORDER FILTER FREQUENCY RESPONSE',//,
C     &' (CASCADED SECOND ORDER SECTIONS)',///)
C FIRST SECOND ORDER STAGE COEFFICIENTS
C   DN213(3)=1.
C   DN213(2)=-1.749875
C   DN213(1)=1.001585
C   DD213(3)=1.
C   DD213(2)=-1.81555
C   DD213(1)=0.990601
C PRINT FIRST STAGE TRANSFER FUNCTION
C   WRITE(4,81)DN213(3),DN213(2),DN213(1),DD213(3),DD213(2),DD213(1)
C   81 FORMAT(' FIRST STAGE TRANSFER FUNCTION (2920 EQUIVALENT)',//,
C     &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/,
C     &' -----',/,
C     &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///)
C SECOND SECOND ORDER STAGE COEFFICIENTS
C   DN223(3)=1.
C   DN223(2)=-1.877805

```

FILE: S221G FORTRAN A1

CALL ENOPL(0)
CALL DONEPL
STOP
END

S2202210
S2202220
S2202230
S2202240

FILE: S221G FORTRAN .A1

CALL CROSS	S2201660
CALL GRAF(0.,100.,1100.,0.0,0.5,2.5)	S2201670
C..... DRAW THE BAR CURVES	S2201680
CALL BARPAT(16)	S2201690
CALL BARWID(0.02)	S2201700
CALL VBARS(XKHZ,Z,YMAG,128)	S2201710
C..... TERMINATE THIS PLOT	S2201720
CALL ENDPL(0)	S2201730
-----	S2201740
C GRAPHICS PARAMETERS FOR YPHASE VS K	S2201750
-----	S2201760
CALL LRGBUF	S2201770
C CALL TEK618	S2201780
CALL COMPRS	S2201790
C..... SETUP THE PLOTTING AREA	S2201800
CALL PAGE (11.0,8.5)	S2201810
CALL NOBRDR	S2201820
CALL AREA2D(9.0,6.5)	S2201830
C..... LABEL THE X & Y AXES	S2201840
CALL XNAME('FREQUENCY (HZ) \$',100)	S2201850
CALL YNAME('PHASE (RADS)\$',100)	S2201860
CALL HEADIN ('DFT OF DIGITAL FILTER IMPULSE RESPONSES',100,1.6,2)	S2201870
CALL HEADIN ('PHASE VS FREQUENCY\$',100,1.,2)	S2201880
C..... DEFINE THE AXES	S2201890
CALL CROSS	S2201900
CALL GRAF(0.,100.,1100.,-2.0,0.5,2.0)	S2201910
C..... DRAW THE PHASE CURVE	S2201920
CALL THKCRV(0.01)	S2201930
CALL MARKER(15)	S2201940
CALL CURVE(XKHZ,YPH,128,0)	S2201950
C..... TERMINATE THIS PLOT	S2201960
CALL ENDPL(0)	S2201970
-----	S2201980
C GRAPHICS PARAMETERS FOR IMPULSE RESPONSE VS N	S2201990
-----	S2202000
CALL LRGBUF	S2202010
C CALL TEK618	S2202020
CALL COMPRS	S2202030
C..... SETUP THE PLOTTING AREA	S2202040
CALL PAGE (11.0,8.5)	S2202050
CALL NOBRDR	S2202060
CALL AREA2D(9.0,6.5)	S2202070
C..... LABEL THE X & Y AXES	S2202080
CALL XNAME('ITERATION (N) \$',100)	S2202090
CALL YNAME('MAGNITUDES',100)	S2202100
CALL HEADIN ('DIGITAL FILTER IMPULSE RESPONSES',100,1.6,3)	S2202110
CALL HEADIN ('OUTPUT VS ITERATION \$',100,1.,3)	S2202120
CALL HEADIN ('INPUT = .017 (SINMAG) AT T = 0\$',100,1.,3)	S2202130
C..... DEFINE THE AXES	S2202140
CALL GRAF(0.,64.,1024.,-.03,.01,.03)	S2202150
C..... DRAW THE IMPULSE RESPONSE CURVE	S2202160
CALL THKCRV(0.01)	S2202170
CALL MARKER(15)	S2202180
CALL CURVE(XK,XN,1024,0)	S2202190
C..... TERMINATE THIS PLOT	S2202200


```

C PRINT PARAMETERS FOR EACH SIMULATION ITERATION
C*****
      IM1=I-1
      WRITE(4,100)IM1,TX,X1(1),Y2(1)
100    FORMAT(1X,I4,2X,3(F13.6,2X))
C USE THIS OUTPUT FORMAT FOR EASYPLOT ROUTINE
C      WRITE(4,100)TX,X1(1),Y2(1)
C 100    FORMAT(1X,3(F15.8,2X))
C*****
C PERFORM SIMULATION SHIFT DELAY
      DO 15 J=2,3,1
        Y1(5-J)=Y1(4-J)
        X1(5-J)=X1(4-J)
        Y2(5-J)=Y2(4-J)
        X2(5-J)=X2(4-J)
15     CONTINUE
10     CONTINUE
-20    CONTINUE
C THIS PROGRAM CALCULATES THE DFT OF THE IMPULSE RESPONSE
C OVER 1024 VALUES. THIS IMPLIES M = 9 (2**9=1024). IWK IS AN
C INTEGER VECTOR FOR FFT2C CALCULATION OF LENGTH M+1 = 10
      M=10
      DO 30 I=1,1024
        A(I)=CONJG(A(I))
30     CONTINUE
      CALL FFT2C(A,M,IWK)
      WRITE(4,38)
38     FORMAT(///,1X,'DFT IMPULSE RESPONSE OUTPUT OVER 1024 ITERATIONS'
& ,/, '      K          YMAG          YPHASE',/)
      DO 40 I=1,1024
        Z(I-1)=0.
        A(I)=CONJG(A(I))
        XK(I)=FLOAT(I-1)
        XKHZ(I)=(1./T)*(XK(I)/1024.)
        YMAG(I)=CABS(A(I))
        YPH(I)=ATAN(AIMAG(A(I))/REAL(A(I)))
        WRITE(4,39)XK(I),YMAG(I),YPH(I)
39     FORMAT(1X,F8.0,4X,F12.4,4X,F12.4)
40     CONTINUE
C-----
C GRAPHICS PARAMETERS FOR YMAG VS K
C-----
      CALL LRGBUF
C      CALL TEK618
      CALL COMPRS
C .....   SETUP THE PLOTTING AREA
      CALL PAGE (11.0,8.5)
      CALL NOBRDR
      CALL AREA2D(9.0,6.5)
C .....   LABEL THE X & Y AXES
      CALL XNAME('FREQUENCY (HZ)$',100)
      CALL YNAME('MAGNITUDES',100)
      CALL HEADIN ('DFT OF DIGITAL FILTER IMPULSE RESPONSES',100,1.6,2)
      CALL HEADIN ('MAGNITUDE VS FREQUENCIES',100,1.,2)
C .....   DEFINE THE AXES

```

S2201110
S2201120
S2201130
S2201140
S2201150
S2201160
S2201170
S2201180
S2201190
S2201200
S2201210
S2201220
S2201230
S2201240
S2201250
S2201260
S2201270
S2201280
S2201290
S2201300
S2201310
S2201320
S2201330
S2201340
S2201350
S2201360
S2201370
S2201380
S2201390
S2201400
S2201410
S2201420
S2201430
S2201440
S2201450
S2201460
S2201470
S2201480
S2201490
S2201500
S2201510
S2201520
S2201530
S2201540
S2201550
S2201560
S2201570
S2201580
S2201590
S2201600
S2201610
S2201620
S2201630
S2201640
S2201650

FILE: S221G FORTRAN A1

```

      SCALS1=0.
      SCALS2=0.
C SAMPLE PERIOD = T = 1.1521152 X 10**-4 SECONDS
      T=4.*192./6.666E6
C COMPUTE SCALE SUM FOR STAGES TO LIMIT INPUT AMPLITUDE
      DO 6 K=1,3
        SCALS1=SCALS1+ABS(DN213(K))
        SCALS1=SCALS1+ABS(DD213(K))
        SCALS2=SCALS2+ABS(DN223(K))
        SCALS2=SCALS2+ABS(DD223(K))
      6 CONTINUE
C ENSURE SCALE SUM FACTORS WILL LIMIT OUTPUT TO LESS THAN ONE
      SCALS1=SCALS1+1.E-6
      SCALS2=SCALS2+1.E-6
C COMPUTE SIMULATED INPUT MAGNITUDE LIMIT
      SINMAG=1./(SCALS1*SCALS2)
C PRINT STAGE SCALE SUMS AND INPUT MAGNITUDE LIMIT
      WRITE(4,85)SCALS1,SCALS2,SINMAG
      85 FORMAT(//,'SCALE FACTORS AND INPUT MAGNITUDE LIMIT',//,
        &'FIRST STAGE SCALE SUM = ',E14.6,/,
        &'SECOND STAGE SCALE SUM = ',E14.6,/,
        &'INPUT AMPLITUDE LIMITED TO +/- ',E14.6,///)
C PERFORM IMPULSE RESPONSE SIMULATION
C INITIALIZE STAGE INPUTS AND OUTPUTS
      DO 5 I=1,3
        X1(I)=0.
        X2(I)=0.
        Y1(I)=0.
        Y2(I)=0.
      5 CONTINUE
C PRINT SIMULATION HEADINGS
      WRITE(4,98)
      98 FORMAT(///,'FILTER IMPULSE RESPONSE',//)
      WRITE(4,99)
      99 FORMAT(/,' I                      IN1                      OUT2',//)
C COMPUTE SIMULATED FILTER RESPONSE OVER INDICATED NUMBER OF SAMPLES (I)
C*****
C ADJUST AS NECESSARY
      DO 10 I=1,1024
C*****
C IN1 = X1(1) = IMPULSE INPUT AT T=0.
      IF (I.EQ.1) X1(1)=SINMAG
      IF (I.NE.1) X1(1)=0.
C TX = TOTAL ELAPSED SAMPLE TIME
      TX=T*FLOAT(I-1)
C OUT1 = Y1(1) = FIRST STAGE OUTPUT = IN2 = X2(1) = SECOND STAGE INPUT
      Y1(1)=DN213(3)*X1(1)+DN213(2)*X1(2)+DN213(1)*X1(3)-
      & DD213(2)*Y1(2)-DD213(1)*Y1(3)
      X2(1)=Y1(1)
C OUT2 = Y2(1) = SECOND STAGE OUTPUT = FILTER OUTPUT
      Y2(1)=DN223(3)*X2(1)+DN223(2)*X2(2)+DN223(1)*X2(3)-
      & DD223(2)*Y2(2)-DD223(1)*Y2(3)
      XN(1)=Y2(1)
C FORM COMPLEX ARRAY OF IMPULSE VALUES FOR DFT
      A(I)=CMPLX(Y2(1),0.)

```

S2200560
S2200570
S2200580
S2200590
S2200600
S2200610
S2200620
S2200630
S2200640
S2200650
S2200660
S2200670
S2200680
S2200690
S2200700
S2200710
S2200720
S2200730
S2200740
S2200750
S2200760
S2200770
S2200780
S2200790
S2200800
S2200810
S2200820
S2200830
S2200840
S2200850
S2200860
S2200870
S2200880
S2200890
S2200900
S2200910
S2200920
S2200930
S2200940
S2200950
S2200960
S2200970
S2200980
S2200990
S2201000
S2201010
S2201020
S2201030
S2201040
S2201050
S2201060
S2201070
S2201080
S2201090
S2201100

```

C*****S2200010
C
C      THIS PROGRAM SIMULATES THE EXECUTION OF A 2920 PROGRAM      *S2200020
C      FOR A 600 HZ BANDPASS FILTER TRANSFER FUNCTION WHICH IS THE *S2200030
C      PRODUCT OF TWO SECOND ORDER FILTER SECTIONS                *S2200040
C                                                                    *S2200050
C                                                                    *S2200060
C      
$$H(Z) = \frac{Y(Z)}{X(Z)} = \frac{DN213(Z)}{DD213(Z)} \times \frac{DN223(Z)}{DD223(Z)}$$
 *S2200070
C                                                                    *S2200080
C                                                                    *S2200090
C      THE SIMULATION IS PERFORMED FOR AN IMPULSE INPUT EQUAL TO THE *S2200100
C      GREATEST ALLOWABLE INPUT (SINMAG) AT T=0. WE THEN EXAMINE *S2200120
C      THE FAST FOURIER TRANSFORM OF THE IMPULSE SEQUENCE OVER 1024 *S2200130
C      SAMPLES TO CONFIRM THE FREQUENCY RESPONSE OF THE SYSTEM.    *S2200140
C                                                                    *S2200150
C*****S2200160
C VARIABLE DECLARATIONS
C   INTEGER IM1,M,IWK(11)                                          S2200170
C   REAL X1(3),X2(3),Y1(3),Y2(3)                                  S2200180
C   REAL SCALS1,SCALS2,SINMAG                                      S2200190
C   REAL DN213(3),DD213(3),DN223(3),DD223(3),T,TX                S2200200
C   COMPLEX A(1024)                                                S2200210
C   REAL XK(1024),XKHZ(1024),XN(1024),YMG(1024),YPH(1024),Z(1024) S2200220
C   PRINT OUTPUT HEADING                                           S2200230
C   WRITE(4,80)                                                     S2200240
C   80 FORMAT('EIGHTH ORDER FILTER IMPULSE RESPONSE',/,           S2200250
C   &' (CASCADED SECOND ORDER SECTIONS)',/,                         S2200260
C   &'(SECOND FOURTH ORDER BLOCK OF EIGHTH ORDER FILTER)',//)     S2200270
C FIRST SECOND ORDER STAGE COEFFICIENTS                            S2200280
C   DN213(3)=1.                                                     S2200290
C   DN213(2)=-1.749875                                              S2200300
C   DN213(1)=1.001585                                              S2200310
C   DD213(3)=1.                                                     S2200320
C   DD213(2)=-1.81555                                              S2200330
C   DD213(1)=0.990601                                              S2200340
C   PRINT FIRST STAGE TRANSFER FUNCTION                             S2200350
C   WRITE(4,81)DN213(3),DN213(2),DN213(1),DD213(3),DD213(2),DD213(1) S2200360
C   81 FORMAT('SECOND FIRST STAGE TRANSFER FUNCTION',/,           S2200370
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/,           S2200380
C   &' -----',/,                                               S2200390
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///)         S2200400
C SECOND SECOND ORDER STAGE COEFFICIENTS                           S2200410
C   DN223(3)=1.                                                     S2200420
C   DN223(2)=-1.877805                                              S2200430
C   DN223(1)=0.998169                                              S2200440
C   DD223(3)=1.                                                     S2200450
C   DD223(2)=-1.809446                                              S2200460
C   DD223(1)=0.982544                                              S2200470
C   PRINT SECOND STAGE TRANSFER FUNCTION                             S2200480
C   WRITE(4,82)DN223(3),DN223(2),DN223(1),DD223(3),DD223(2),DD223(1) S2200490
C   82 FORMAT('SECOND SECOND STAGE TRANSFER FUNCTION',/,           S2200500
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/,           S2200510
C   &' -----',/,                                               S2200520
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///)         S2200530
C INITIALIZE VARIABLES                                             S2200540
C                                                                    S2200550

```

S221 PROGRAM OUTPUT

FOURTH ORDER FILTER IMPULSE RESPONSE
(CASCADED SECOND ORDER SECTIONS)

FIRST STAGE TRANSFER FUNCTION (2920 COEFFICIENTS)

$$\begin{array}{l} 0.100000E+01 + -0.174988E+01 Z^{-1} + 0.100159E+01 Z^{-2} \\ \hline 0.100000E+01 + -0.181555E+01 Z^{-1} + 0.990601E+00 Z^{-2} \end{array}$$

SECOND STAGE TRANSFER FUNCTION (2920 COEFFICIENTS)

$$\begin{array}{l} 0.100000E+01 + -0.187780E+01 Z^{-1} + 0.998169E+00 Z^{-2} \\ \hline 0.100000E+01 + -0.180945E+01 Z^{-1} + 0.982554E+00 Z^{-2} \end{array}$$

SCALE FACTORS AND INPUT MAGNITUDE LIMIT

FIRST STAGE SCALE SUM = 0.755761E+01
SECOND STAGE SCALE SUM = 0.766797E+01
INPUT AMPLITUDE LIMITED TO +/- 0.172558E-01

FILTER IMPULSE RESPONSE

MAX AMPLITUDE OF IMPULSE RESPONSE OVER 512 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 1024 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 1536 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 2048 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 2560 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 3072 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 3584 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 4096 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 4608 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 5120 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 5632 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 6144 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 6656 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 7168 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 7680 ITERATIONS = 0.011935
MAX AMPLITUDE OF IMPULSE RESPONSE OVER 8192 ITERATIONS = 0.011935

MAX AMPLITUDE OCCURRED AT ITERATION 150

FILE: S221 FORTRAN A1

& D2(2,2)*Y2(2)-D2(2,1)*Y2(3)	S2201110
IF ((MXAMP.LT.(ABS(Y2(1))))).AND.(1.NE.1)) GO TO 30	S2201120
GO TO 40	S2201130
30 MXAMP=ABS(Y2(1))	S2201140
IMXAMP=1	S2201150
40 CONTINUE	S2201160
C PRINT PARAMETERS FOR EACH SIMULATION ITERATION	S2201170
C*****	S2201180
C WRITE(4,100)IM1,TX,X1(1),Y2(1)	S2201190
C 100 FORMAT(1X,14,2X,3(F13.6,2X))	S2201200
C USE THIS OUTPUT FORMAT FOR EASYPLOT ROUTINE	S2201210
C WRITE(4,100)TX,X1(1),Y2(1)	S2201220
C 100 FORMAT(1X,3(F15.8,2X))	S2201230
C*****	S2201240
C PERFORM SIMULATION SHIFT DELAY	S2201250
DO 15 J=2,3,1	S2201260
Y1(5-J)=Y1(4-J)	S2201270
X1(5-J)=X1(4-J)	S2201280
Y2(5-J)=Y2(4-J)	S2201290
X2(5-J)=X2(4-J)	S2201300
15 CONTINUE	S2201310
IF (MOD(1,512))10,14,10	S2201320
14 WRITE(4,16)1,MXAMP	S2201330
16 FORMAT(' MAX AMPLITUDE OF IMPULSE RESPONSE OVER ',15,	S2201340
&' ITERATIONS = ',F9.6)	S2201350
10 CONTINUE	S2201360
WRITE(4,18)IMXAMP	S2201370
18 FORMAT(/,' MAX AMPLITUDE OCCURRED AT ITERATION ',15)	S2201380
STOP	S2201390
END	S2201400

```

      82 FORMAT(' SECOND STAGE TRANSFER FUNCTION (2920 COEFFICIENTS)',//, S2200560
        &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/, S2200570
        &' -----',/, S2200580
        &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///) S2200590
C INITIALIZE VARIABLES S2200600
  SCALS1=0. S2200610
  SCALS2=0. S2200620
C SAMPLE PERIOD = T = 1.1521152 X 10**-4 SECONDS S2200630
  T=4.*192./6.666E6 S2200640
C COMPUTE SCALE SUM FOR STAGES TO LIMIT INPUT AMPLITUDE S2200650
  DO 6 K=1,3 S2200660
    SCALS1=SCALS1+ABS(N2(1,K)) S2200670
    SCALS1=SCALS1+ABS(D2(1,K)) S2200680
    SCALS2=SCALS2+ABS(N2(2,K)) S2200690
    SCALS2=SCALS2+ABS(D2(2,K)) S2200700
  6 CONTINUE S2200710
C ENSURE SCALE SUM FACTORS WILL LIMIT OUTPUT TO LESS THAN ONE S2200720
  SCALS1=SCALS1+1.E-6 S2200730
  SCALS2=SCALS2+1.E-6 S2200740
C COMPUTE SIMULATED INPUT MAGNITUDE LIMIT S2200750
  SINMAG=1./(SCALS1*SCALS2) S2200760
C PRINT STAGE SCALE SUMS AND INPUT MAGNITUDE LIMIT S2200770
  WRITE(4,85)SCALS1,SCALS2,SINMAG S2200780
  85 FORMAT(//,' SCALE FACTORS AND INPUT MAGNITUDE LIMIT',///, S2200790
    &' FIRST STAGE SCALE SUM = ',E14.6,/, S2200800
    &' SECOND STAGE SCALE SUM = ',E14.6,/, S2200810
    &' INPUT AMPLITUDE LIMITED TO +/- ',E14.6,///) S2200820
C PERFORM IMPULSE RESPONSE SIMULATION S2200830
C INITIALIZE STAGE INPUTS AND OUTPUTS S2200840
  DO 5 I=1,3 S2200850
    X1(I)=0. S2200860
    X2(I)=0. S2200870
    Y1(I)=0. S2200880
    Y2(I)=0. S2200890
  5 CONTINUE S2200900
  MXAMP=0. S2200910
  IMXAMP=0 S2200920
C PRINT SIMULATION HEADINGS S2200930
  WRITE(4,98) S2200940
  98 FORMAT(' FILTER IMPULSE RESPONSE',//) S2200950
C WRITE(4,99) S2200960
C 99 FORMAT(/,' I TIME IN1 OUT2',//) S2200970
C COMPUTE SIMULATED FILTER RESPONSE OVER REQUIRED ITERATIONS (NUMIT) S2200980
  DO 10 I=1,NUMIT S2200990
C IN1 = X1(1) = IMPULSE INPUT AT T=0. S2201000
  IF (I.EQ.1) X1(1)=SINMAG S2201010
  IF (I.NE.1) X1(1)=0. S2201020
C TX = TOTAL ELAPSED SAMPLE TIME S2201030
  TX=T*FLOAT(I-1) S2201040
C OUT1 = Y1(1) = FIRST STAGE OUTPUT = IN2 = X2(1) = SECOND STAGE INPUT S2201050
  Y1(1)=N2(1,3)*X1(1)+N2(1,2)*X1(2)+N2(1,1)*X1(3)- S2201060
  & D2(1,2)*Y1(2)-D2(1,1)*Y1(3) S2201070
  X2(1)=Y1(1) S2201080
C OUT2 = Y2(1) = SECOND STAGE OUTPUT = XN(1) S2201090
  Y2(1)=N2(2,3)*X2(1)+N2(2,2)*X2(2)+N2(2,1)*X2(3)- S2201100

```

```

C*****S2200010
C
C                                APPENDIX F                                S2200020
C
C                                FORTRAN PROGRAM S221                        S2200030
C
C                                THIS PROGRAM SIMULATES THE EXECUTION OF A 2920 PROGRAM S2200040
C                                FOR A FOURTH ORDER ELLIPTIC 600 HZ BANDPASS FILTER TRANSFER S2200050
C                                TRANSFER FUNCTION WHICH IS THE PRODUCT OF TWO SECOND ORDER S2200060
C                                FILTER SECTIONS S2200070
C
C                                
$$H(Z) = \frac{Y(Z)}{X(Z)} = \frac{N2(1,Z)}{D2(1,Z)} \times \frac{N2(2,Z)}{D2(2,Z)}$$
 S2200080
C                                THE SIMULATION IS PERFORMED FOR AN IMPULSE INPUT EQUAL TO THE S2200090
C                                GREATEST ALLOWABLE INPUT (SINMAG) AT T=0 AND ALLOWED TO RUN S2200100
C                                OVER NUMIT ITERATIONS TO CHECK FOR STABILITY. S2200110
C
C*****S2200120
C
C                                S2200130
C                                S2200140
C                                S2200150
C                                S2200160
C                                S2200170
C                                S2200180
C                                S2200190
C                                S2200200
C*****S2200210
C VARIABLE DECLARATIONS S2200220
C   INTEGER NUMIT, IMXAMP S2200230
C   REAL X1(3), X2(3), Y1(3), Y2(3) S2200240
C   REAL SCALS1, SCALS2, SINMAG, MXAMP S2200250
C   REAL N2(2,3), D2(2,3), T, TX S2200260
C DECLARE NUMBER OF SIMULATION ITERATIONS S2200270
C   NUMIT=8192 S2200280
C PRINT OUTPUT HEADING S2200290
C   WRITE(4,80) S2200300
C   80 FORMAT(' S221 PROGRAM OUTPUT',//, S2200310
C   &' FOURTH ORDER FILTER IMPULSE RESPONSE',//, S2200320
C   &' (CASCADED SECOND ORDER SECTIONS)',///) S2200330
C FIRST SECOND ORDER STAGE COEFFICIENTS S2200340
C   N2(1,3)=1. S2200350
C   N2(1,2)=-1.749875 S2200360
C   N2(1,1)=1.001585 S2200370
C   D2(1,3)=1. S2200380
C   D2(1,2)=-1.81555 S2200390
C   D2(1,1)=0.990601 S2200400
C PRINT FIRST STAGE TRANSFER FUNCTION S2200410
C   WRITE(4,81)N2(1,3),N2(1,2),N2(1,1),D2(1,3),D2(1,2),D2(1,1) S2200420
C   81 FORMAT(' FIRST STAGE TRANSFER FUNCTION (2920 COEFFICIENTS)',//, S2200430
C   &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/, S2200440
C   &' -----',/, S2200450
C   &' ',E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///) S2200460
C SECOND SECOND ORDER STAGE COEFFICIENTS S2200470
C   N2(2,3)=1. S2200480
C   N2(2,2)=-1.877805 S2200490
C   N2(2,1)=0.998169 S2200500
C   D2(2,3)=1. S2200510
C   D2(2,2)=-1.809446 S2200520
C   D2(2,1)=0.982554 S2200530
C PRINT SECOND STAGE TRANSFER FUNCTION S2200540
C   WRITE(4,82)N2(2,3),N2(2,2),N2(2,1),D2(2,3),D2(2,2),D2(2,1) S2200550

```

FILE: DBPFR FORTRAN A1

CALL PAGE (11.0,8.5)	DBP01110
CALL NOBRDR	DBP01120
CALL AREA2D(9.0,6.5)	DBP01130
C LABEL THE X & Y AXES	DBP01140
CALL XNAME('FREQUENCY (HZ)\$',100)	DBP01150
CALL YNAME('PHASE (DEGREES)\$',100)	DBP01160
CALL HEADIN('DIGITAL ELLIPTIC BPF PHASE RESPONSES',100,1.6,2)	DBP01170
CALL HEADIN('PHASE (DEGREES) VS FREQ (FO=590 HZ)\$',100,1.,2)	DBP01180
C DEFINE THE AXES	DBP01190
CALL GRAF(0.0,'SCALE',1200.,-100.,'SCALE',100.)	DBP01200
C DRAW THE CURVES	DBP01210
CALL THKCRV(0.02)	DBP01220
CALL MARKER(15)	DBP01230
CALL CURVE(FREQ,HPHASE,201,0)	DBP01240
C TERMINATE THIS PLOT	DBP01250
CALL ENDPL(0)	DBP01260
CALL DONEPL	DBP01270
STOP	DBP01280
END	DBP01290

C GRAPHICS PARAMETERS FOR MAGNITUDE VS FREQUENCY (IN HZ)	DBP00560
C-----	DBP00570
CALL LRGBUF	DBP00580
C CALL COMPRS	DBP00590
CALL TEK618	DBP00600
C CALL VRSTEC(0,0,0)	DBP00610
C SETUP THE PLOTTING AREA	DBP00620
CALL PAGE (11.0,8.5)	DBP00630
CALL NOBRDR	DBP00640
CALL AREA2D(9.0,6.5)	DBP00650
C LABEL THE X & Y AXES	DBP00660
CALL XNAME('FREQUENCY (HZ)\$',100)	DBP00670
CALL YNAME('AMPLITUDES',100)	DBP00680
CALL HEADIN('DIGITAL ELLIPTIC BPF FREQUENCY RESPONSES',100,1.6,2)	DBP00690
CALL HEADIN('NORMALIZED AMPLITUDE VS FREQ (FO=590 HZ)\$',100,1.,2)	DBP00700
C DEFINE THE AXES	DBP00710
CALL GRAF(0.0,'SCALE',1200.,-0.5,'SCALE',1.5)	DBP00720
C DRAW THE CURVES	DBP00730
CALL THKCRV(0.02)	DBP00740
CALL MARKER(15)	DBP00750
CALL CURVE(FREQ,HMAGN,201,0)	DBP00760
C TERMINATE THIS PLOT	DBP00770
CALL ENDPL(0)	DBP00780
C-----	DBP00790
C GRAPHICS PARAMETERS FOR MAGNITUDE (IN DB) VS FREQUENCY (IN HZ)	DBP00800
C-----	DBP00810
CALL LRGBUF	DBP00820
C CALL COMPRS	DBP00830
CALL TEK618	DBP00840
C CALL VRSTEC(0,0,0)	DBP00850
C SETUP THE PLOTTING AREA	DBP00860
CALL PAGE (11.0,8.5)	DBP00870
CALL NOBRDR	DBP00880
CALL AREA2D(9.0,6.5)	DBP00890
C LABEL THE X & Y AXES	DBP00900
CALL XNAME('FREQUENCY (HZ)\$',100)	DBP00910
CALL YNAME('AMPLITUDE (DB)\$',100)	DBP00920
CALL HEADIN('DIGITAL ELLIPTIC BPF FREQUENCY RESPONSES',100,1.6,2)	DBP00930
CALL HEADIN('AMPLITUDE (DB) VS FREQ (FO=590 HZ)\$',100,1.,2)	DBP00940
C DEFINE THE AXES	DBP00950
CALL GRAF(0.0,'SCALE',1200.,20.0,'SCALE',10.0)	DBP00960
C DRAW THE CURVES	DBP00970
CALL THKCRV(0.02)	DBP00980
CALL MARKER(15)	DBP00990
CALL CURVE(FREQ,HMAGDB,201,0)	DBP01000
C TERMINATE THIS PLOT	DBP01010
CALL ENDPL(0)	DBP01020
C-----	DBP01030
C GRAPHICS PARAMETERS FOR PHASE VS FREQUENCY	DBP01040
C-----	DBP01050
CALL LRGBUF	DBP01060
CALL COMPRS	DBP01070
C CALL TEK618	DBP01080
C CALL VRSTEC(0,0,0)	DBP01090
C SETUP THE PLOTTING AREA	DBP01100

```

C*****DBP00010
C                                         *DBP00020
C                                         *DBP00030
C                                         *DBP00040
C                                         *DBP00050
C                                         *DBP00060
C                                         *DBP00070
C                                         *DBP00080
C                                         *DBP00090
C                                         *DBP00100
C                                         *DBP00110
C                                         DBP00120
C                                         DBP00130
C                                         DBP00140
C                                         DBP00150
C                                         DBP00160
C                                         DBP00170
C                                         DBP00180
C                                         DBP00190
C                                         DBP00200
C                                         DBP00210
C                                         DBP00220
C                                         DBP00230
C                                         DBP00240
C                                         DBP00250
C                                         DBP00260
C                                         DBP00270
C                                         DBP00280
C                                         DBP00290
C                                         DBP00300
C                                         DBP00310
C                                         DBP00320
C                                         DBP00330
C                                         DBP00340
C                                         DBP00350
C                                         DBP00360
C                                         DBP00370
C                                         DBP00380
C                                         DBP00390
C                                         DBP00400
C                                         DBP00410
C                                         DBP00420
C                                         DBP00430
C                                         DBP00440
C                                         DBP00450
C                                         DBP00460
C                                         DBP00470
C                                         DBP00480
C                                         DBP00490
C                                         DBP00500
C                                         DBP00510
C                                         DBP00520
C                                         DBP00530
C                                         DBP00540
C-----DBP00550

C                                         APPENDIX E
C                                         FORTRAN PROGRAM DBPFR
C
C PROGRAM TO PLOT DIGITAL BAND-PASS FILTER FREQUENCY AND PHASE
C RESPONSE OF THE ELLIPTIC FILTER TRANSFER FUNCTION
C
C TYPE DECLARATIONS
C   IMPLICIT REAL(A-H,O-Z), INTEGER(I-N)
C   REAL OMEGA(201), HMAG(201), HPHASE(201), HMAGN(201), HMAGDB(201)
C   REAL F(201), FREQ(201), FS, FSDIV2, TS
C   COMPLEX Z, H
C
C   NORMALIZED TRANSFER FUNCTION COEFFICIENTS
C     A0 = 1.
C     A1 = -3.6279
C     A2 = 5.2861
C     A3 = A2
C     A4 = A1
C     B0 = 1.
C     B1 = -3.6251
C     B2 = 5.2586
C     B3 = -3.5768
C     B4 = 0.97353
C
C   CONSTANTS
C     PI = 3.1415927
C     FS = 6.666E6/(4.*192.)
C     TS = 1./FS
C     FSDIV2 = FS/2.
C
C   EVALUATE MAGNITUDE AND PHASE OF H(EXP(J*OMEGA*T))
C     DO 10 I = 1, 201
C       F(I) = FLOAT(I-1)
C       FREQ(I) = 6.*F(I)
C       OMEGA(I) = (2.*PI.*FREQ(I)*TS)
C       Z = CMPLX(COS(OMEGA(I)), SIN(OMEGA(I)))
C       H = (A0+A1*Z**(-1)+A2*Z**(-2)+A3*Z**(-3)+A4*Z**(-4))/(B0+
C &B1*Z**(-1)+B2*Z**(-2)+B3*Z**(-3)+B4*Z**(-4))
C       HMAG(I) = CABS(H)
C       X = REAL(H)
C       Y = AIMAG(H)
C       HPHASE(I) = ATAN(Y/X)*180./PI
C     10 CONTINUE
C
C   NORMALIZE MAGNITUDE
C     AMAX = 0.0
C     DO 20 I = 1, 201
C       IF(HMAG(I).GT.AMAX) AMAX = HMAG(I)
C     20 CONTINUE
C     DO 30 I = 1, 201
C       HMAGN(I) = HMAG(I)/AMAX
C       HMAGDB(I) = 20.0 * ALOG10(HMAG(I))
C     30 CONTINUE
C-----

```

FILE: S22F FORTRAN A1

```

C          WRITE(4,99)                                     S2201110
C 99      FORMAT(/,' I      TIME      IN1      OUT1=IN2      OUT2',//) S2201120
C COMPUTE SIMULATED FILTER RESPONSE OVER INDICATED NUMBER OF SAMPLES (I) S2201130
C*****                                               S2201140
C ADJUST AS NECESSARY                                     S2201150
C          DO 10 I=1,2048                                   S2201160
C*****                                               S2201170
C TX = TOTAL ELAPSED SAMPLE TIME                         S2201180
C          TX=T*FLOAT(I-1)                                 S2201190
C IN1 = X1(1) = FILTER FIRST STAGE INPUT VALUE (LIMITED BY SINMAG) S2201200
C          X1(1)=SINMAG*SIN(TWOPIF*TX)                   S2201210
C OUT1 = Y1(1) = FIRST STAGE OUTPUT = IN2 = X2(1) = SECOND STAGE INPUT S2201220
C          Y1(1)=DN213(3)*X1(1)+DN213(2)*X1(2)+DN213(1)*X1(3)- S2201230
C          & DD213(2)*Y1(2)-DD213(1)*Y1(3)                S2201240
C          X2(1)=Y1(1)                                     S2201250
C OUT2 = Y2(1) = SECOND STAGE OUTPUT = FILTER OUTPUT     S2201260
C          Y2(1)=DN223(3)*X2(1)+DN223(2)*X2(2)+DN223(1)*X2(3)- S2201270
C          & DD223(2)*Y2(2)-DD223(1)*Y2(3)                S2201280
C PRINT PARAMETERS FOR EACH SIMULATION ITERATION         S2201290
C*****                                               S2201300
C          WRITE(4,100)I,TX,X1(1),X2(1),Y2(1)            S2201310
C 100      FORMAT(1X,14,2X,4(F10.3,2X))                  S2201320
C USE THIS OUTPUT FORMAT FOR EASYPLOT ROUTINE            S2201330
C          WRITE(4,100)TX,X1(1),Y2(1)                    S2201340
C 100      FORMAT(1X,3(F15.8,2X))                         S2201350
C*****                                               S2201360
C REMEMBER MAXIMUM AMPLITUDE IN EACH FREQUENCY SIMULATION TRIAL S2201370
C          IF (ABS(Y2(1))-MOUT) 11,11,14                  S2201380
C          14      MOUT=ABS(Y2(1))                         S2201390
C          IMOUT=I                                         S2201400
C PERFORM SIMULATION SHIFT DELAY                         S2201410
C          11      DO 15 J=2,3,1                           S2201420
C          Y1(5-J)=Y1(4-J)                                S2201430
C          X1(5-J)=X1(4-J)                                S2201440
C          Y2(5-J)=Y2(4-J)                                S2201450
C          X2(5-J)=X2(4-J)                                S2201460
C          15      CONTINUE                                S2201470
C          10      CONTINUE                                S2201480
C PRINT MAXIMUM OUTPUT AMPLITUDE FOR EACH FREQUENCY SIMULATION RUN S2201490
C          WRITE(4,89)MOUT,IMOUT                           S2201500
C          89      FORMAT(' MAXIMUM OUTPUT AMPLITUDE = ',F15.8,/, S2201510
C          &' THIS OCCURRED AT SIMULATION ITERATION ',15,///) S2201520
C          20 CONTINUE                                     S2201530
C          STOP                                             S2201540
C          END                                             S2201550

```

PROGRAM S22F OUTPUT

FOURTH ORDER FILTER FREQUENCY RESPONSE
(CASCADED SECOND ORDER SECTIONS)

FIRST STAGE TRANSFER FUNCTION (2920 EQUIVALENT)

$$\begin{array}{l} 0.100000E+01 + -0.174988E+01 Z^{*-1} + 0.100159E+01 Z^{*-2} \\ \hline 0.100000E+01 + -0.181555E+01 Z^{*-1} + 0.990601E+00 Z^{*-2} \end{array}$$

SECOND STAGE TRANSFER FUNCTION (2920 EQUIVALENT)

$$\begin{array}{l} 0.100000E+01 + -0.187780E+01 Z^{*-1} + 0.998169E+00 Z^{*-2} \\ \hline 0.100000E+01 + -0.180945E+01 Z^{*-1} + 0.982544E+00 Z^{*-2} \end{array}$$

SCALE FACTORS AND INPUT MAGNITUDE LIMIT

FIRST STAGE SCALE SUM = 0.755761E+01
SECOND STAGE SCALE SUM = 0.766796E+01
INPUT AMPLITUDE LIMITED TO +/- 0.172558E-01

FILTER FREQUENCY RESPONSE FOR F = 500. HZ

MAXIMUM OUTPUT AMPLITUDE = 0.10085225
THIS OCCURRED AT SIMULATION ITERATION 180

FILTER FREQUENCY RESPONSE FOR F = 575. HZ

MAXIMUM OUTPUT AMPLITUDE = 1.59343243
THIS OCCURRED AT SIMULATION ITERATION 515

FILTER FREQUENCY RESPONSE FOR F = 591. HZ

MAXIMUM OUTPUT AMPLITUDE = 1.62147427
THIS OCCURRED AT SIMULATION ITERATION 500

FILTER FREQUENCY RESPONSE FOR F = 600. HZ

MAXIMUM OUTPUT AMPLITUDE = 0.89969766
THIS OCCURRED AT SIMULATION ITERATION 284

FILTER FREQUENCY RESPONSE FOR F = 625. HZ

MAXIMUM OUTPUT AMPLITUDE = 0.30743510
THIS OCCURRED AT SIMULATION ITERATION 135

FILTER FREQUENCY RESPONSE FOR F = 700. HZ

MAXIMUM OUTPUT AMPLITUDE = 0.07697707
THIS OCCURRED AT SIMULATION ITERATION 150

```

C*****S2200010
C                                             *S2200020
C   THIS PROGRAM SIMULATES THE EXECUTION OF A 2920 PROGRAM *S2200030
C   FOR A 600 HZ BANDPASS FILTER TRANSFER FUNCTION WHICH IS THE *S2200040
C   PRODUCT OF TWO SECOND ORDER FILTER SECTIONS *S2200050
C                                             *S2200060
C   
$$H(Z) = \frac{Y(Z)}{X(Z)} = \frac{DN213(Z)}{DD213(Z)} \times \frac{DN223(Z)}{DD223(Z)}$$
 *S2200070
C                                             *S2200080
C   THE SIMULATION IS PERFORMED OVER A RANGE OF DIFFERENT INPUT *S2200090
C   FREQUENCIES ABOUT THE TARGET CENTER FREQUENCY OF 600 HZ. *S2200100
C   AFTER SIMULATION THE FREQUENCY RESPONSE IS PLOTTED FOR *S2200110
C   GRAPHICAL REVIEW. *S2200120
C                                             *S2200130
C*****S2200140
C   VARIABLE DECLARATIONS *S2200150
C   INTEGER IMOUT *S2200160
C   REAL TX,F(9) *S2200170
C   REAL NUM(1024) *S2200180
C   REAL IN(1024) *S2200190
C   REAL OUT(1024) *S2200200
C   REAL X1(3),X2(3),Y1(3),Y2(3) *S2200210
C   REAL SCALS1,SCALS2,SINMAG,MOUT *S2200220
C   REAL DN213(3),DD213(3),DN223(3),DD223(3),T,PI,TWOPIF *S2200230
C INPUT FREQUENCIES *S2200240
C   F(1)=700 *S2200250
C   F(2)= *S2200260
C   F(3)= *S2200270
C   F(4)= *S2200280
C   F(5)= *S2200290
C   F(6)= *S2200300
C   F(7)= *S2200310
C   F(8)= *S2200320
C   F(9)= *S2200330
C   WRITE(4,80) *S2200340
C   80 FORMAT('FOURTH ORDER FILTER FREQUENCY RESPONSE',/, *S2200350
C   &' (CASCADED SECOND ORDER SECTIONS)',//) *S2200360
C FIRST SECOND ORDER STAGE COEFFICIENTS *S2200370
C   DN213(3)=1. *S2200380
C   DN213(2)=-1.749875 *S2200390
C   DN213(1)=1.001585 *S2200400
C   DD213(3)=1. *S2200410
C   DD213(2)=-1.81555 *S2200420
C   DD213(1)=0.990601 *S2200430
C PRINT FIRST STAGE TRANSFER FUNCTION *S2200440
C   WRITE(4,81)DN213(3),DN213(2),DN213(1),DD213(3),DD213(2),DD213(1) *S2200450
C   81 FORMAT('FIRST STAGE TRANSFER FUNCTION (2920 EQUIVALENT)',/, *S2200460
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/, *S2200470
C   &'-----',/, *S2200480
C   &'E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///) *S2200490
C SECOND SECOND ORDER STAGE COEFFICIENTS *S2200500
C   DN223(3)=1. *S2200510
C   DN223(2)=-1.877805 *S2200520
C   DN223(1)=0.998169 *S2200530
C   *S2200540
C   *S2200550

```

```

DD223(3)=1.
DD223(2)=-1.809446
DD223(1)=0.982544
C PRINT SECOND STAGE TRANSFER FUNCTION
WRITE(4,82)DN223(3),DN223(2),DN223(1),DD223(3),DD223(2),DD223(1)
82 FORMAT('SECOND STAGE TRANSFER FUNCTION (2920 EQUIVALENT)',//,
&E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',/,
&'-----',/,
&E14.6,' + ',E14.6,' Z**-1 + ',E14.6,' Z**-2',///)
C INITIALIZE VARIABLES
PI=3.1415927
SCALS1=0.
SCALS2=0.
C SAMPLE PERIOD = T = 1.1521152 X 10**-4 SECONDS
T=4.*192./6.666E6
C COMPUTE SCALE SUM FOR STAGES TO LIMIT INPUT AMPLITUDE
DO 6 K=1,3
SCALS1=SCALS1+ABS(DN213(K))
SCALS1=SCALS1+ABS(DD213(K))
SCALS2=SCALS2+ABS(DN223(K))
SCALS2=SCALS2+ABS(DD223(K))
6 CONTINUE
C ENSURE SCALE SUM FACTORS WILL LIMIT OUTPUT TO LESS THAN ONE
SCALS1=SCALS1+1.E-6
SCALS2=SCALS2+1.E-6
C COMPUTE SIMULATED INPUT MAGNITUDE LIMIT
SINMAG=1./(SCALS1*SCALS2)
C PRINT STAGE SCALE SUMS AND INPUT MAGNITUDE LIMIT
WRITE(4,85)SCALS1,SCALS2,SINMAG
85 FORMAT(//,'SCALE FACTORS AND INPUT MAGNITUDE LIMIT',//,
&'FIRST STAGE SCALE SUM = ',E14.6,/,
&'SECOND STAGE SCALE SUM = ',E14.6,/,
&'INPUT AMPLITUDE LIMITED TO +/- ',E14.6,///)
C BEGIN SIMULATION FOR SPECIFIED FREQUENCIES GIVEN BY F(L)
C*****
C ADJUST AS NECESSARY
DO 20 L=1,1
C*****
C COMPUTE SIMULATION RUN INPUT CONSTANT FOR EACH FREQUENCY
TWOPIF=2.*PI*F(L)
C INITIALIZE STAGE INPUTS AND OUTPUTS
IMOUT=0
MOUT=0.
DO 5 I=1,3
X1(I)=0.
X2(I)=0.
Y1(I)=0.
Y2(I)=0.
5 CONTINUE
C PRINT SIMULATION HEADINGS
WRITE(4,98)F(L)
98 FORMAT(///,'FILTER FREQUENCY RESPONSE FOR F = ',F5.0,' HZ',//)
WRITE(4,99)
99 FORMAT(/,' I      TIME      IN1      OUT1=IN2      OUT2',//)
C COMPUTE SIMULATED FILTER RESPONSE OVER INDICATED NUMBER OF SAMPLES (I)

```

S2200560
S2200570
S2200580
S2200590
S2200600
S2200610
S2200620
S2200630
S2200640
S2200650
S2200660
S2200670
S2200680
S2200690
S2200700
S2200710
S2200720
S2200730
S2200740
S2200750
S2200760
S2200770
S2200780
S2200790
S2200800
S2200810
S2200820
S2200830
S2200840
S2200850
S2200860
S2200870
S2200880
S2200890
S2200900
S2200910
S2200920
S2200930
S2200940
S2200950
S2200960
S2200970
S2200980
S2200990
S2201000
S2201010
S2201020
S2201030
S2201040
S2201050
S2201060
S2201070
S2201080
S2201090
S2201100

FILE: S22FG FORTRAN A1

```

C*****
C ADJUST AS NECESSARY
DO 10 I=1,1024
C*****
C TX = TOTAL ELAPSED SAMPLE TIME
TX=T*FLOAT(I-1)
C IN1 = X1(1) = FILTER FIRST STAGE INPUT VALUE (LIMITED BY SINMAG)
X1(1)=SINMAG*SIN(TWOPIF*TX)
NUM(1)=FLOAT(1)
IN(1)=X1(1)
C OUT1 = Y1(1) = FIRST STAGE OUTPUT = IN2 = X2(1) = SECOND STAGE INPUT
Y1(1)=DN213(3)*X1(1)+DN213(2)*X1(2)+DN213(1)*X1(3)-
& DD213(2)*Y1(2)-DD213(1)*Y1(3)
X2(1)=Y1(1)
C OUT2 = Y2(1) = SECOND STAGE OUTPUT = FILTER OUTPUT
Y2(1)=DN223(3)*X2(1)+DN223(2)*X2(2)+DN223(1)*X2(3)-
& DD223(2)*Y2(2)-DD223(1)*Y2(3)
OUT(1)=Y2(1)
C PRINT PARAMETERS FOR EACH SIMULATION ITERATION
C*****
C WRITE(4,100)I,TX,X1(1),X2(1),Y2(1)
C 100 FORMAT(1X,14,2X,4(F10.3,2X))
C USE THIS OUTPUT FORMAT FOR EASYPLOT ROUTINE
C WRITE(4,100)TX,X1(1),Y2(1)
C 100 FORMAT(1X,3(F15.8,2X))
C*****
C REMEMBER MAXIMUM AMPLITUDE IN EACH FREQUENCY SIMULATION TRIAL
IF (ABS(Y2(1))-MOUT) 11,11,14
14 MOUT=ABS(Y2(1))
IMOUT=I
C PERFORM SIMULATION SHIFT DELAY
11 DO 15 J=2,3,1
Y1(5-J)=Y1(4-J)
X1(5-J)=X1(4-J)
Y2(5-J)=Y2(4-J)
X2(5-J)=X2(4-J)
15 CONTINUE
10 CONTINUE
C PRINT MAXIMUM OUTPUT AMPLITUDE FOR EACH FREQUENCY SIMULATION RUN
WRITE(4,89)F(L),MOUT,IMOUT
89 FORMAT(' MAXIMUM OUTPUT AMPLITUDE FOR ',F5.0,' HZ = ',F15.8,/,
&' THIS OCCURRED AT SIMULATION ITERATION ',15)
20 CONTINUE
C-----
C GRAPHICS PARAMETERS FOR FREQUENCY RESPONSE OUTPUT VS INPUT
C-----
CALL LRGBUF
CALL TEK618
CALL COMPRS
C ..... SETUP THE PLOTTING AREA
CALL PAGE (11.0,8.5)
CALL NOBRDR
CALL AREA2D(9.0,6.5)
C ..... LABEL THE X & Y AXES
CALL XNAME(' ITERATION (N) S',100)

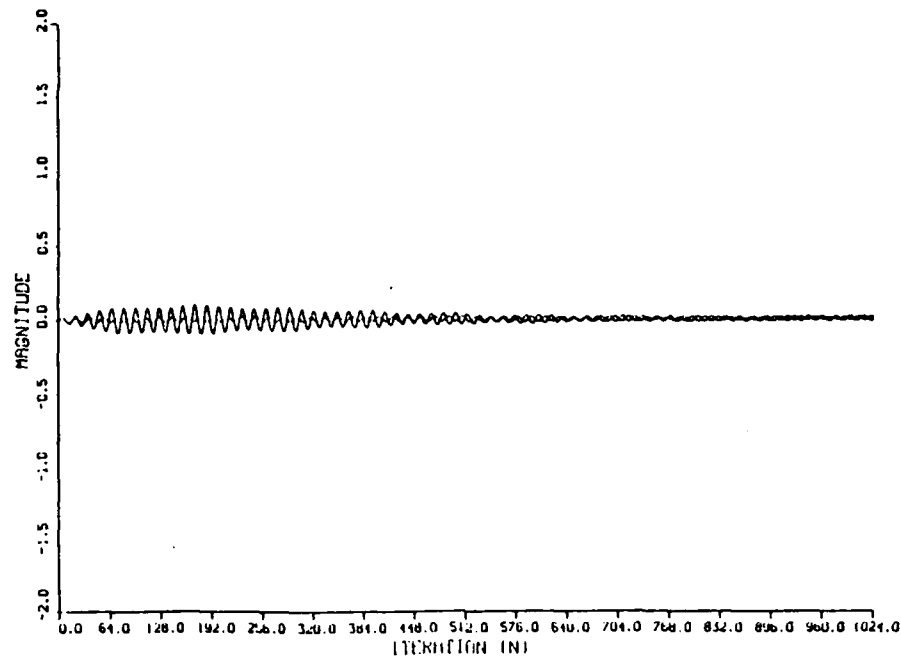
```

S2201110
S2201120
S2201130
S2201140
S2201150
S2201160
S2201170
S2201180
S2201190
S2201200
S2201210
S2201220
S2201230
S2201240
S2201250
S2201260
S2201270
S2201280
S2201290
S2201300
S2201310
S2201320
S2201330
S2201340
S2201350
S2201360
S2201370
S2201380
S2201390
S2201400
S2201410
S2201420
S2201430
S2201440
S2201450
S2201460
S2201470
S2201480
S2201490
S2201500
S2201510
S2201520
S2201530
S2201540
S2201550
S2201560
S2201570
S2201580
S2201590
S2201600
S2201610
S2201620
S2201630
S2201640
S2201650

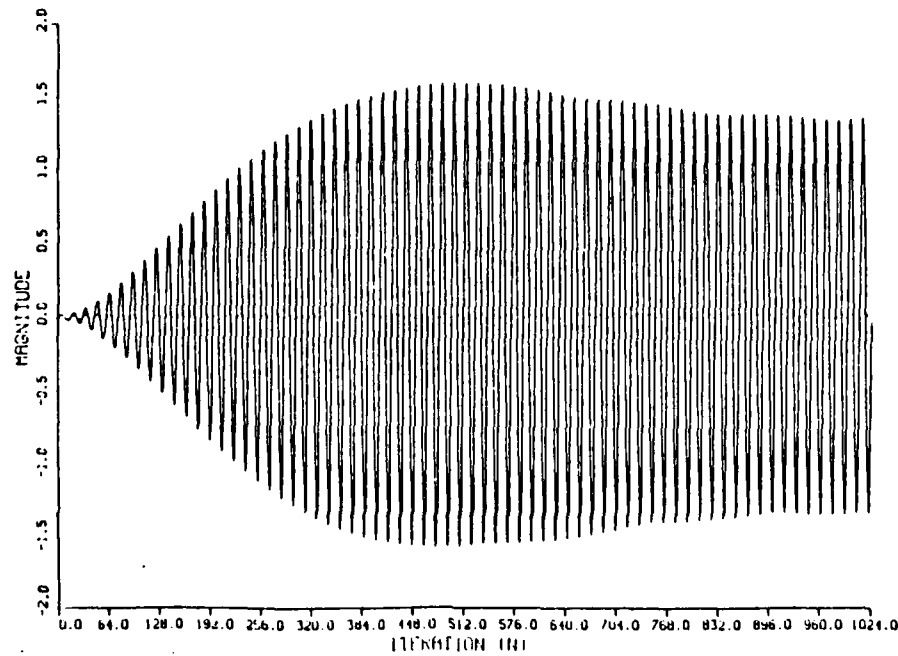
FILE: S22FG FORTRAN A1

CALL YNAME('MAGNITUDES',100)	S2201660
CALL HEADIN ('DIGITAL FILTER FREQUENCY RESPONSES',100,1.6,3)	S2201670
CALL HEADIN ('SIMULATION INPUT/OUTPUT VS ITERATIONS',100,1.,3)	S2201680
CALL HEADIN ('FREQUENCY = 700 HZS',100,1.,3)	S2201690
C DEFINE THE AXES	S2201700
CALL GRAF(0.,64.,1024.,-2.0,.5,2.0)	S2201710
C DRAW THE INPUT CURVE	S2201720
CALL THKCRV(0.01)	S2201730
CALL MARKER(15)	S2201740
CALL CURVE(NUM,IN,1024,0)	S2201750
C DRAW THE OUTPUT CURVE	S2201760
CALL THKCRV(0.01)	S2201770
CALL MARKER(15)	S2201780
CALL CURVE(NUM,OUT,1024,0)	S2201790
C TERMINATE THIS PLOT	S2201800
CALL ENDPL(0)	S2201810
CALL DONEPL	S2201820
STOP	S2201830
END	S2201840

DIGITAL FILTER FREQUENCY RESPONSE SIMULATION INPUT/OUTPUT VS ITERATION FREQUENCY = 500 HZ



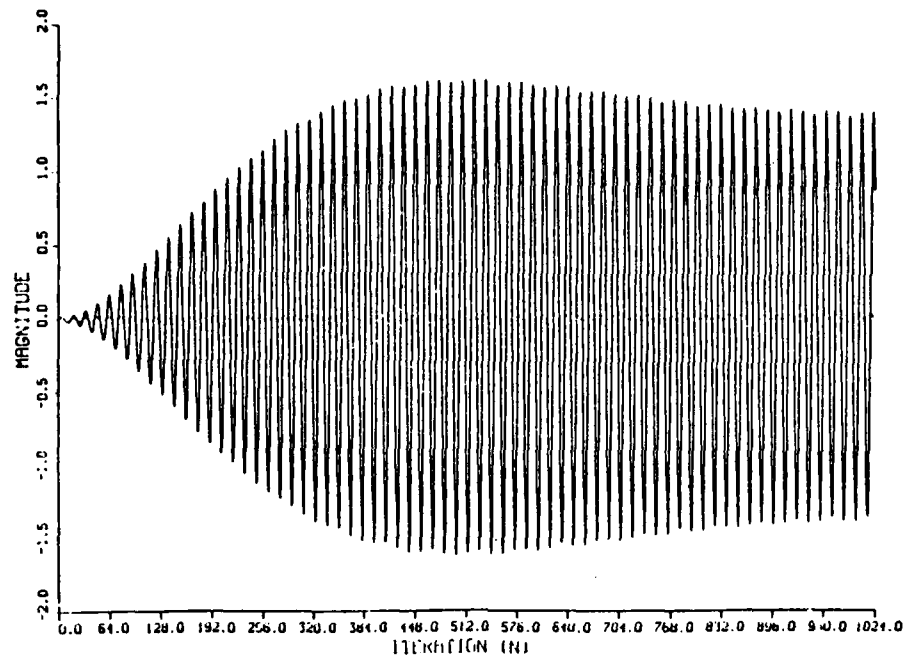
DIGITAL FILTER FREQUENCY RESPONSE SIMULATION INPUT/OUTPUT VS ITERATION FREQUENCY = 575 HZ



DIGITAL FILTER FREQUENCY RESPONSE

SIMULATION INPUT/OUTPUT VS ITERATION

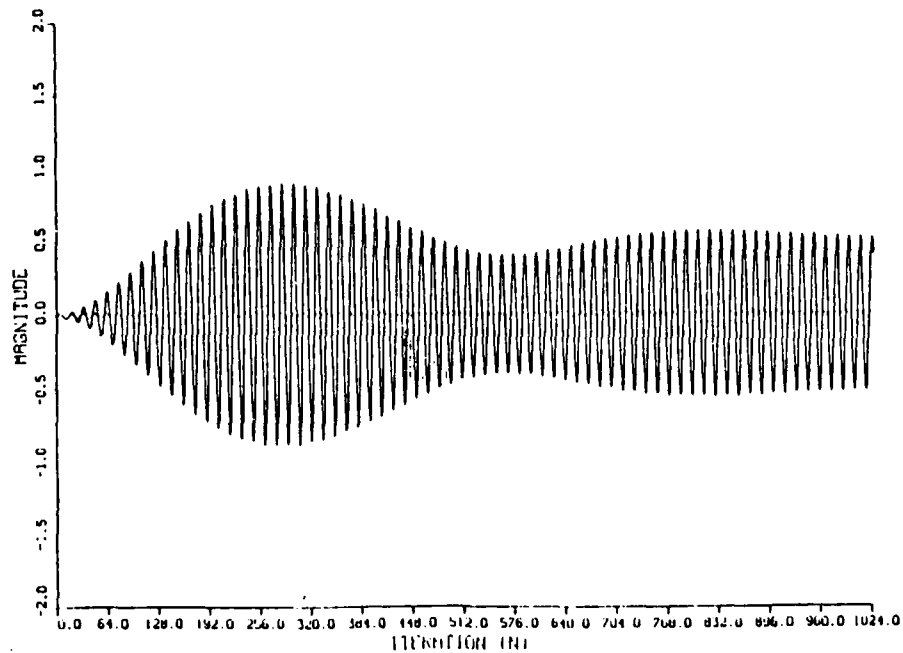
FREQUENCY = 590.825 HZ



DIGITAL FILTER FREQUENCY RESPONSE

SIMULATION INPUT/OUTPUT VS ITERATION

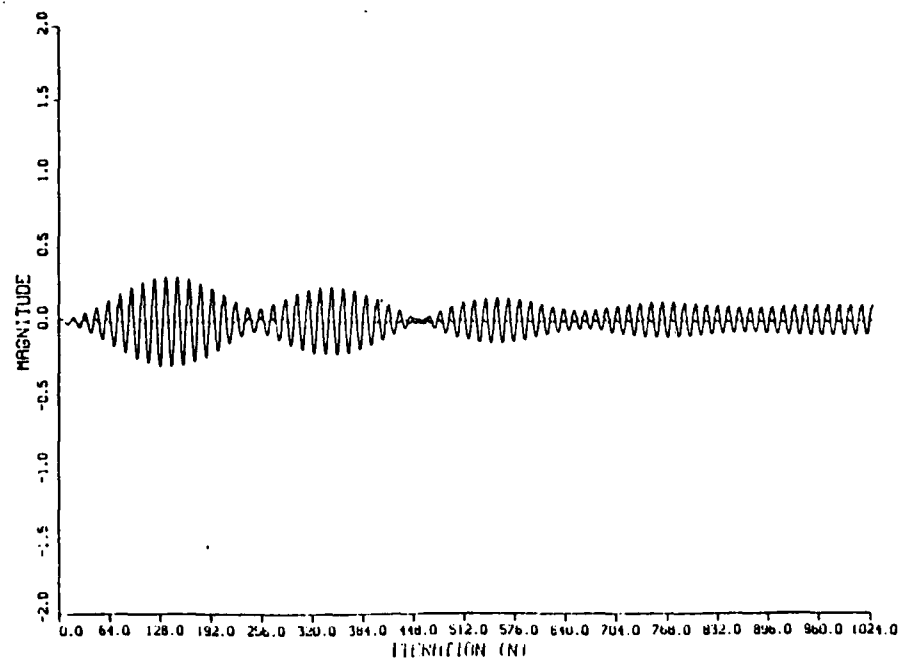
FREQUENCY = 600 HZ



DIGITAL FILTER FREQUENCY RESPONSE

SIMULATION INPUT/OUTPUT VS ITERATION

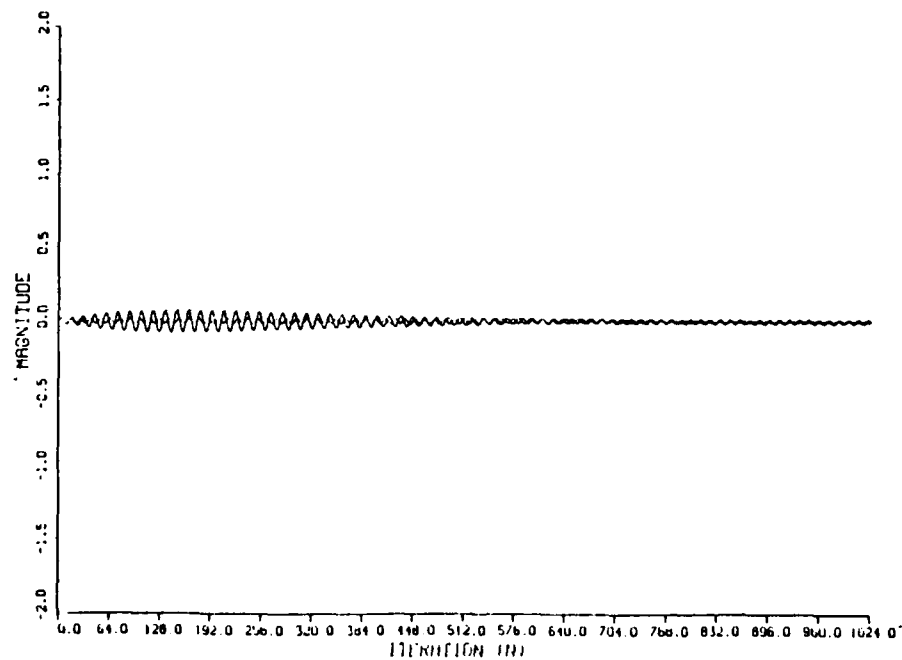
FREQUENCY = 625 HZ



DIGITAL FILTER FREQUENCY RESPONSE

SIMULATION INPUT/OUTPUT VS ITERATION

FREQUENCY = 700 HZ



ASSEMBLER INVOKED BY: PS2920 SM344

LINE LOC OBJECT SOURCE STATEMENT

```

1      :
2      :
3      :
4      :
5      :
6      :
7      :
8      :
9      :
10     0 4000EF : CLEAR IAP REGISTER
11     :
12     :
13     :
14     1 0000EF : IN0
15     2 0000EF : IN0
16     3 0000EF : IN0
17     4 0000EF : IN0
18     5 4000EF : NOP
19     6 4000EF : NOP
20     7 4000EF : NOP
21     :
22     :
23     :
24     8 6000EF : OUT5
25     9 4000EF : NOP
26    10 4000EF : NOP
27    11 4000EF : NOP
28    12 7100EF : OUT7
29    13 4000EF : NOP
30    14 4000EF : NOP
31    15 4000EF : NOP
32    16 6100EF : OUT6
33    17 4000EF : NOP
34    18 4000EF : NOP
35    19 4000EF : NOP
36    20 5100EF : OUT5
37    21 4000EF : NOP
38    22 4000EF : NOP
39    23 4000EF : NOP
40    24 4100EF : OUT4
41    25 4000EF : NOP
42    26 4000EF : NOP
43    27 4000EF : NOP
44    28 3100EF : OUT3
45    29 4000EF : NOP
46    30 4000EF : NOP
47    31 4000EF : NOP
48    32 2100EF : OUT2
49    33 4000EF : NOP
50    34 4000EF : NOP
51    35 4000EF : NOP
52    36 1100EF : OUT1

```

LINE	LOC	OBJECT	SOURCE STATEMENT
53	37	4000EF	NOP
54	38	4000EF	NOP
55	39	4000EF	NOP
56	40	0100EF	OUT0
57	41	4000EF	NOP
58	42	4000EF	NOP
59	43	4000EF	NOP
60			:
61			: SCALE DOWN DIGITAL INPUT BY A FACTOR OF 64
62	44	4006AE	LDA IAR,IAR,R06
63			:
64			: LOAD SCALED INPUT FROM IAR INTO X11
65	45	4022EF	LIA X11,IAR
66			:
67			: INITIALIZE Y11
68	46	4003FF	LDA Y11,KP0
69			:
70			: PERFORM FIRST STAGE DIFFERENCE EQUATION COMPUTATION
71			: $Y11=N13*X11-N12*X12+N11*X13-D13*Y12-D11*Y13$
72			:
73			: $Y11=N13*X11$
74			: $(N13=1.000000)$
75			:
76	47	4400EF	LDA FRD,X11
77			:
78	48	4300FD	ADD Y11,FRD
79			:
80			: $Y11=Y11-N12*X12$
81			: $(N12=1.749875)$
82			:
83	49	4600EF	LDA FRD,X12,R00
84	50	460000	ADD FRD,X12,R01
85	51	460040	ADD FRD,X12,R02
86	52	460060	ADD FRD,X12,R04
87	53	460080	ADD FRD,X12,R05
88	54	460090	ADD FRD,X12,R06
89	55	4600C0	ADD FRD,X12,R07
90	56	4600E0	ADD FRD,X12,R08
91	57	46000D	ADD FRD,X12,R09
92	58	46002D	ADD FRD,X12,R10
93	59	46004D	ADD FRD,X12,R11
94	60	46006D	ADD FRD,X12,R12
95	61	46008D	ADD FRD,X12,R13
96			:
97	62	4200FB	SUB Y11,FRD
98			:
99			: $Y11=Y11+N11*Y13$
100			: $(N11=1.001585)$
101			:
102	63	4000EF	LDA FRD,X13,R00
103	64	40008D	ADD FRD,X13,R10
104	65	40004D	ADD FRD,X13,R11
105	66	40008D	ADD FRD,X13,R13
106			:

LINE	LOC	OBJECT	SOURCE STATEMENT
107	67	4200FD	ADD Y11,RPD
108			:
109			: Y11=Y11+D12*Y12
110			: (D12=1.815550)
111			:
112	68	4008EF	LDA RPD,Y12,R00
113	69	400800	ADD RPD,Y12,R01
114	70	400820	ADD RPD,Y12,R02
115	71	400860	ADD RPD,Y12,R04
116	72	400800	ADD RPD,Y12,R09
117	73	400820	ADD RPD,Y12,R10
118	74	400880	ADD RPD,Y12,R13
119			:
120	75	4200FD	ADD Y11,RPD
121			:
122			: Y11=Y11-D11*Y13
123			: (D11=2.990601)
124			:
125	76	4E000E	LDA RPD,Y13,R01
126	77	4E0020	ADD RPD,Y13,R02
127	78	4E0040	ADD RPD,Y13,R03
128	79	4E0060	ADD RPD,Y13,R04
129	80	4E0080	ADD RPD,Y13,R05
130	81	4E00A0	ADD RPD,Y13,R06
131	82	4E00C0	ADD RPD,Y13,R08
132	83	4E0000	ADD RPD,Y13,R09
133	84	4E0060	ADD RPD,Y13,R12
134	85	4E0080	ADD RPD,Y13,R13
135			:
136	86	4200FB	SUB Y11,RPD
137			:
138			: INITIALIZE Y21
139	87	4498FF	LDA Y21,RP0
140			:
141			: PERFORM SECOND STAGE DIFFERENCE EQUATION COMPUTATION
142			: Y21=N23*Y11-N22*Y22+N21*Y23+I22*Y22-I21*Y23
143			:
144			: Y21=N23*Y11
145			: (N23=1.000000)
146			:
147	88	4408EF	LDA RPD,Y11
148			:
149	89	4610FD	ADD Y21,RPD
150			:
151			: Y21=Y21-N22*Y22
152			: (N22=1.877005)
153			:
154	90	4420EF	LDA RPD,Y22,R00
155	91	442000	ADD RPD,Y22,R01
156	92	442020	ADD RPD,Y22,R02
157	93	442040	ADD RPD,Y22,R03
158	94	442000	ADD RPD,Y22,R09
159	95	442040	ADD RPD,Y22,R11
160	96	442060	ADD RPD,Y22,R12

LINE	LOC	OBJECT	SOURCE STATEMENT
161	97	44200D	ADD FRD,Y22,R13
162			:
163	98	4610FB	SUB Y21,FRD
164			:
165			: Y21=Y21+M21*Y23
166			: (M21=0.998169)
167			:
168	99	44200E	LDA FRD,Y23,R01
169	100	44202C	ADD FRD,Y23,R02
170	101	44204C	ADD FRD,Y23,R03
171	102	44206C	ADD FRD,Y23,R04
172	103	44208C	ADD FRD,Y23,R05
173	104	4420AC	ADD FRD,Y23,R06
174	105	4420CC	ADD FRD,Y23,R07
175	106	4420EC	ADD FRD,Y23,R08
176	107	44200D	ADD FRD,Y23,R09
177	108	44208D	ADD FRD,Y23,R13
178			:
179	109	4610FD	ADD Y21,FRD
180			:
181			: Y21=Y21-ID22*Y22
182			: (ID22=1.809446)
183			:
184	110	4620EF	LDA FRD,Y22,R00
185	111	46200C	ADD FRD,Y22,R01
186	112	46202C	ADD FRD,Y22,R02
187	113	46204C	ADD FRD,Y22,R03
188	114	46206C	ADD FRD,Y22,R04
189	115	46208C	ADD FRD,Y22,R05
190	116	4620AC	ADD FRD,Y22,R06
191	117	4620CC	ADD FRD,Y22,R07
192	118	4620EC	ADD FRD,Y22,R08
193	119	46200D	ADD FRD,Y22,R09
194			:
195	120	4610FD	ADD Y21,FRD
196			:
197			: Y21=Y21-ID21*Y23
198			: (ID21=0.982544)
199			:
200	121	46200E	LDA FRD,Y23,R01
201	122	46202C	ADD FRD,Y23,R02
202	123	46204C	ADD FRD,Y23,R03
203	124	46206C	ADD FRD,Y23,R04
204	125	46208C	ADD FRD,Y23,R05
205	126	4620CC	ADD FRD,Y23,R07
206	127	4620EC	ADD FRD,Y23,R08
207	128	46200D	ADD FRD,Y23,R09
208	129	46208D	ADD FRD,Y23,R13
209			:
210	130	4610FB	SUB Y21,FRD
211			:
212			: LOAD Y21 INTO DAF FOR OUTPUT
213	131	4A40EF	LDA DAF,Y21,R00
214			:

AD-A158 960 A MATCHED FILTER ALGORITHM FOR ACOUSTIC SIGNAL
DETECTION(U) NAVAL POSTGRADUATE SCHOOL MONTEREY CA
D W JORDAN JUN 85

AD-A158 960 A MATCHED FILTER ALGORITHM FOR ACOUSTIC SIGNAL
DETECTION(U) NAVAL POSTGRADUATE SCHOOL MONTEREY CA
D W JORDAN JUN 85

AD-A158 960 A MATCHED FILTER ALGORITHM FOR ACOUSTIC SIGNAL DETECTION(U) NAVAL POSTGRADUATE SCHOOL MONTEREY CA D W JORDAN JUN 85 3/3

UNCLASSIFIED F/G 17/1

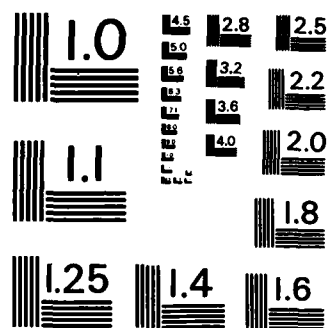
UNCLASSIFIED F/G 17/1

UNCLASSIFIED F/G 17/1 NL

FILMED

FILMED

OTIC



MICROCOPY RESOLUTION TEST CHART
NATIONAL BUREAU OF STANDARDS-1963-A

LINE LOC OBJECT SOURCE STATEMENT

```

215          ; MULTIPLY OUTPUT BY A FACTOR OF 4
216 132 4066AF LDA DAR,DAR,L02
217 133 4000EF NOP
218 134 4000EF NOP
219 135 4000EF NOP
220 136 4000EF NOP
221 137 4000EF NOP
222 138 4000EF NOP
223          ;
224          ; OUTPUT VALUE IN DAR TO CHANNEL 0
225 139 8000EF OUT0
226 140 8000EF OUT0
227 141 8000EF OUT0
228 142 8000EF OUT0
229          ;
230          ; SERIAL REGISTER SHIFT FOR NEXT PROGRAM PASS
231 143 4660FF LDA Y03,Y02,R00
232 144 4E48EF LDA Y02,Y01,R00
233 145 4C18EF LDA Y13,Y12,R00
234 146 4018FF LDA Y12,Y11,R00
235 147 4060FF LDA X03,X02,R00
236 148 4048EF LDA X02,Y11,R00
237 149 4218EF LDA X13,X12,R00
238 150 4400FF LDA X12,X11,R00
239 151 4000EF NOP
240 152 4000EF NOP
241 153 4000EF NOP
242 154 4000EF NOP
243 155 4000EF NOP
244 156 4000EF NOP
245 157 4000EF NOP
246 158 4000EF NOP
247 159 4000EF NOP
248 160 4000EF NOP
249 161 4000EF NOP
250 162 4000EF NOP
251 163 4000EF NOP
252 164 4000EF NOP
253 165 4000EF NOP
254 166 4000EF NOP
255 167 4000EF NOP
256 168 4000EF NOP
257 169 4000EF NOP
258 170 4000EF NOP
259 171 4000EF NOP
260 172 4000EF NOP
261 173 4000EF NOP
262 174 4000EF NOP
263 175 4000EF NOP
264 176 4000EF NOP
265 177 4000EF NOP
266 178 4000EF NOP
267 179 4000EF NOP
268 180 4000EF NOP

```

LINE	LOC	OBJECT	SOURCE	STATEMENT
269	181	4000EF		NOP
270	182	4000EF		NOP
271	183	4000EF		NOP
272	184	4000EF		NOP
273	185	4000EF		NOP
274	186	4000EF		NOP
275	187	4000EF		NOP
276				
277				; THIS IS THE FINAL FOUR INSTRUCTION SEGMENT
278	188	5000EF		EOP
279	189	4000EF		NOP
280	190	4000EF		NOP
281	191	4000EF		NOP
282				END

SYMBOL:

VALUE:

X11	0
Y11	1
PPD	0
X12	0
X13	4
Y12	15
Y13	15
Y21	3
X22	0
X23	0
Y22	10
Y23	11

ASSEMBLY COMPLETE

ERRORS	=	0
WARNINGS	=	0
RAMSIZE	=	12
ROMSIZE	=	192

```

C*****CTR00010
C                                     *CTR00020
C                                     *CTR00030
C                                     *CTR00040
C                                     *CTR00050
C                                     *CTR00060
C                                     *CTR00070
C                                     *CTR00080
C                                     *CTR00090
C                                     *CTR00100
C                                     *CTR00110
C VARIABLE DECLARATIONS CTR00120
C   REAL N2(2,3),D2(2,3),N2TX,D2TX,TRIAL,TWOVAL,FKM1 CTR00130
C   REAL N2B(2,3),D2B(2,3),N2P(2,3),D2P(2,3) CTR00140
C   INTEGER N2T(2,3,14),D2T(2,3,14) CTR00150
C   INTEGER JM1,JX,KM1 CTR00160
C ACTUAL FIRST STAGE COEFFICIENTS TO BE TRANSFORMED CTR00170
C   N2(1,3)=1. CTR00180
C   N2(1,2)=-1.7499 CTR00190
C   N2(1,1)=1.0017 CTR00200
C   D2(1,3)=1. CTR00210
C   D2(1,2)=-1.8156 CTR00220
C   D2(1,1)=0.99068 CTR00230
C ACTUAL SECOND STAGE COEFFICIENTS TO BE TRANSFORMED CTR00240
C   N2(2,3)=1. CTR00250
C   N2(2,2)=-1.8779 CTR00260
C   N2(2,1)=0.99817 CTR00270
C   D2(2,3)=1. CTR00280
C   D2(2,2)=-1.8095 CTR00290
C   D2(2,1)=0.98255 CTR00300
C INITIALIZE BINARY COEFFICIENT MATRIX CTR00310
C   DO 10 I=1,2 CTR00320
C     DO 12 J=1,3 CTR00330
C       N2B(I,J)=0. CTR00340
C       D2B(I,J)=0. CTR00350
C       N2P(I,J)=0. CTR00360
C       D2P(I,J)=0. CTR00370
C       DO 14 K=1,13 CTR00380
C         N2T(I,J,K)=0 CTR00390
C         D2T(I,J,K)=0 CTR00400
C       CONTINUE CTR00410
C     CONTINUE CTR00420
C   CONTINUE CTR00430
C PERFORM COEFFICIENT TRANSFORMATION TO BINARY 2920 REPRESENTATION CTR00440
C NUMERATOR TERMS CTR00450
C   DO 20 I=1,2 CTR00460
C     DO 22 J=1,3 CTR00470
C       N2TX=ABS(N2(I,J)) CTR00480
C       IF (N2TX-1.0) 221,222,223 CTR00490
222       N2T(I,J,1)=1 CTR00500
C       N2B(I,J)=1.0 CTR00510
C       GO TO 22 CTR00520
223       N2T(I,J,1)=1 CTR00530
C       N2B(I,J)=1.0 CTR00540
C       N2TX=N2TX-1. CTR00550

```

FILE: CTRANS2 FORTRAN A1

221	DO 24 K=2, 14	CTR00560
	FKM1=FLOAT(K-1)	CTR00570
	TWOVAL=1./(2.**FKM1)	CTR00580
	TRIAL=N2TX-TWOVAL	CTR00590
	IF (TRIAL) 25, 242, 243	CTR00600
242	N2T(I, J, K)=1	CTR00610
	N2B(I, J)=N2B(I, J)+TWOVAL	CTR00620
	GO TO 20	CTR00630
243	N2T(I, J, K)=1	CTR00640
	N2B(I, J)=N2B(I, J)+TWOVAL	CTR00650
	N2TX=TRIAL	CTR00660
	GO TO 24	CTR00670
25	IF (K.EQ.14) N2T(I, J, K)=0	CTR00680
24	CONTINUE	CTR00690
22	CONTINUE	CTR00700
20	CONTINUE	CTR00710
C	DENOMINATOR TERMS	CTR00720
	DO 30 I=1, 2	CTR00730
	DO 32 J=1, 3	CTR00740
	D2TX=ABS(D2(I, J))	CTR00750
	IF (D2TX-1.0) 321, 322, 323	CTR00760
322	D2T(I, J, 1)=1	CTR00770
	D2B(I, J)=1.0	CTR00780
	GO TO 32	CTR00790
323	D2T(I, J, 1)=1	CTR00800
	D2B(I, J)=1.0	CTR00810
	D2TX=D2TX-1.	CTR00820
321	DO 34 K=2, 14	CTR00830
	FKM1=FLOAT(K-1)	CTR00840
	TWOVAL=1./(2.**FKM1)	CTR00850
	TRIAL=D2TX-TWOVAL	CTR00860
	IF (TRIAL) 35, 342, 343	CTR00870
342	D2T(I, J, K)=1	CTR00880
	D2B(I, J)=D2B(I, J)+TWOVAL	CTR00890
	GO TO 30	CTR00900
343	D2T(I, J, K)=1	CTR00910
	D2B(I, J)=D2B(I, J)+TWOVAL	CTR00920
	D2TX=TRIAL	CTR00930
	GO TO 34	CTR00940
35	IF (K.EQ.14) D2T(I, J, K)=0	CTR00950
34	CONTINUE	CTR00960
32	CONTINUE	CTR00970
30	CONTINUE	CTR00980
C	PRINT OUTPUT HEADING	CTR00990
	WRITE(4, 80)	CTR01000
80	FORMAT(' PROGRAM CTRANS2 OUTPUT', //,	CTR01010
	&' FOURTH ORDER DIGITAL FILTER 2920 BINARY EQUIVALENTS', //,	CTR01020
	&' (TWO CASCADED SECOND ORDER SECTIONS)', ///)	CTR01030
C	PRINT FIRST STAGE TRANSFER FUNCTION	CTR01040
	WRITE(4, 81) N2(1, 3), N2(1, 2), N2(1, 1), D2(1, 3), D2(1, 2), D2(1, 1)	CTR01050
81	FORMAT(' FIRST STAGE TRANSFER FUNCTION', //,	CTR01060
	&' ', E14.6, ' + ', E14.6, ' Z**-1 + ', E14.6, ' Z**-2', //,	CTR01070
	&' -----', //,	CTR01080
	&' ', E14.6, ' + ', E14.6, ' Z**-1 + ', E14.6, ' Z**-2')	CTR01090
	WRITE(4, 810)	CTR01100

```

810 FORMAT(///, ' BINARY REPRESENTATION OF NUMERATOR COEFFICIENTS', //, CTR01110
& ' R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13' CTR01120
&, /) CTR01130
DO 811 J=1,3 CTR01140
JX=4-J CTR01150
N2P(1,JX)=N2B(1,JX)/ABS(N2(1,JX)) CTR01160
WRITE(4,812)JX,N2(1,JX) CTR01170
812 FORMAT(' N2(1, ', I1, ') = ', F9.6) CTR01180
WRITE(4,814)N2T(1,JX,1),N2T(1,JX,2),N2T(1,JX,3),N2T(1,JX,4), CTR01190
&N2T(1,JX,5),N2T(1,JX,6),N2T(1,JX,7),N2T(1,JX,8),N2T(1,JX,9), CTR01200
&N2T(1,JX,10),N2T(1,JX,11),N2T(1,JX,12),N2T(1,JX,13),N2T(1,JX,14), CTR01210
&N2B(1,JX),N2P(1,JX) CTR01220
814 FORMAT(9X,14(I1,3X),/, ' ABSOLUTE BINARY EQUIVALENT = ' CTR01230
&, F9.6,/, ' (THIS IS ', F9.6, ' OF THE ACTUAL VALUE)', /) CTR01240
811 CONTINUE CTR01250
WRITE(4,815) CTR01260
815 FORMAT(///, ' BINARY REPRESENTATION OF DENOMINATOR COEFFICIENTS', //, CTR01270
& ' R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13' CTR01280
&, /) CTR01290
DO 816 J=1,3 CTR01300
JX=4-J CTR01310
D2P(1,JX)=D2B(1,JX)/ABS(D2(1,JX)) CTR01320
WRITE(4,817)JX,D2(1,JX) CTR01330
817 FORMAT(' D2(1, ', I1, ') = ', F9.6) CTR01340
WRITE(4,819)D2T(1,JX,1),D2T(1,JX,2),D2T(1,JX,3),D2T(1,JX,4), CTR01350
&D2T(1,JX,5),D2T(1,JX,6),D2T(1,JX,7),D2T(1,JX,8),D2T(1,JX,9), CTR01360
&D2T(1,JX,10),D2T(1,JX,11),D2T(1,JX,12),D2T(1,JX,13),D2T(1,JX,14), CTR01370
&D2B(1,JX),D2P(1,JX) CTR01380
819 FORMAT(9X,14(I1,3X),/, ' ABSOLUTE BINARY EQUIVALENT = ' CTR01390
&, F9.6,/, ' (THIS IS ', F9.6, ' OF THE ACTUAL VALUE)', //) CTR01400
816 CONTINUE CTR01410
C PRINT SECOND STAGE TRANSFER FUNCTION CTR01420
WRITE(4,82)N2(2,3),N2(2,2),N2(2,1),D2(2,3),D2(2,2),D2(2,1) CTR01430
82 FORMAT(' 1' ' SECOND STAGE TRANSFER FUNCTION', //, CTR01440
&E14.6, ' + ', E14.6, ' Z**-1 + ', E14.6, ' Z**-2', //, CTR01450
& ' -----', /, CTR01460
&E14.6, ' + ', E14.6, ' Z**-1 + ', E14.6, ' Z**-2') CTR01470
WRITE(4,820) CTR01480
820 FORMAT(///, ' BINARY REPRESENTATION OF NUMERATOR COEFFICIENTS', //, CTR01490
& ' R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13' CTR01500
&, /) CTR01510
DO 821 J=1,3 CTR01520
JX=4-J CTR01530
N2P(2,JX)=N2B(2,JX)/ABS(N2(2,JX)) CTR01540
WRITE(4,822)JX,N2(2,JX) CTR01550
822 FORMAT(' N2(2, ', I1, ') = ', F9.6) CTR01560
WRITE(4,824)N2T(2,JX,1),N2T(2,JX,2),N2T(2,JX,3),N2T(2,JX,4), CTR01570
&N2T(2,JX,5),N2T(2,JX,6),N2T(2,JX,7),N2T(2,JX,8),N2T(2,JX,9), CTR01580
&N2T(2,JX,10),N2T(2,JX,11),N2T(2,JX,12),N2T(2,JX,13),N2T(2,JX,14), CTR01590
&N2B(2,JX),N2P(2,JX) CTR01600
824 FORMAT(9X,14(I1,3X),/, ' ABSOLUTE BINARY EQUIVALENT = ' CTR01610
&, F9.6,/, ' (THIS IS ', F9.6, ' OF THE ACTUAL VALUE)', //) CTR01620
821 CONTINUE CTR01630
WRITE(4,825) CTR01640
825 FORMAT(///, ' BINARY REPRESENTATION OF DENOMINATOR COEFFICIENTS', //, CTR01650

```

FILE: CTRANS2 FORTRAN A1

```
&'      R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13' CTR01660
&,/)                                         CTR01670
DO 826 J=1,3                               CTR01680
    JX=4-J                                  CTR01690
    D2P(2,JX)=D2B(2,JX)/ABS(D2(2,JX))      CTR01700
    WRITE(4,827)JX,D2(2,JX)                CTR01710
827    FORMAT(' D2(2,'11,') = ',F9.6)      CTR01720
    WRITE(4,829)D2T(2,JX,1),D2T(2,JX,2),D2T(2,JX,3),D2T(2,JX,4), CTR01730
    &D2T(2,JX,5),D2T(2,JX,6),D2T(2,JX,7),D2T(2,JX,8),D2T(2,JX,9), CTR01740
    &D2T(2,JX,10),D2T(2,JX,11),D2T(2,JX,12),D2T(2,JX,13),D2T(2,JX,14), CTR01750
    &D2B(2,JX),D2P(2,JX)                   CTR01760
829    FORMAT(9X,14(11,3X),/, ' ABSOLUTE BINARY EQUIVALENT = ' CTR01770
    &,F9.6,/, ' (THIS IS ',F9.6,' OF THE ACTUAL VALUE)',//) CTR01780
826 CONTINUE                               CTR01790
    STOP                                    CTR01800
    END                                    CTR01810
```


PROGRAM CTRANS2 OUTPUT

FOURTH ORDER DIGITAL FILTER 2920 BINARY EQUIVALENTS
(TWO CASCADED SECOND ORDER SECTIONS)

FIRST STAGE TRANSFER FUNCTION

$$\frac{0.100000E+01 + -0.174990E+01 Z^{-1} + 0.100170E+01 Z^{-2}}{0.100000E+01 + -0.181560E+01 Z^{-1} + 0.990680E+00 Z^{-2}}$$

BINARY REPRESENTATION OF NUMERATOR COEFFICIENTS

R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13

N2(1,3) = 1.000000

1 0 0 0 0 0 0 0 0 0 0 0 0 0

ABSOLUTE BINARY EQUIVALENT = 1.000000
(THIS IS 1.000000 OF THE ACTUAL VALUE)

N2(1,2) = -1.749900

1 1 0 1 1 1 1 1 1 1 1 1 1 1

ABSOLUTE BINARY EQUIVALENT = 1.749875
(THIS IS 0.999986 OF THE ACTUAL VALUE)

N2(1,1) = 1.001700

1 0 0 0 0 0 0 0 0 0 1 1 0 1

ABSOLUTE BINARY EQUIVALENT = 1.001585
(THIS IS 0.999885 OF THE ACTUAL VALUE)

BINARY REPRESENTATION OF DENOMINATOR COEFFICIENTS

R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13

D2(1,3) = 1.000000

1 0 0 0 0 0 0 0 0 0 0 0 0 0

ABSOLUTE BINARY EQUIVALENT = 1.000000
(THIS IS 1.000000 OF THE ACTUAL VALUE)

D2(1,2) = -1.815600

1 1 1 0 1 0 0 0 0 1 1 0 0 1

ABSOLUTE BINARY EQUIVALENT = 1.815550
(THIS IS 0.999972 OF THE ACTUAL VALUE)

D2(1,1) = 0.990680

0 1 1 1 1 1 1 0 1 1 0 0 1 1

ABSOLUTE BINARY EQUIVALENT = 0.990601
(THIS IS 0.999920 OF THE ACTUAL VALUE)

SECOND STAGE TRANSFER FUNCTION

$$\frac{0.100000E+01 + -0.187790E+01 Z^{-1} + 0.998170E+00 Z^{-2}}{0.100000E+01 + -0.180950E+01 Z^{-1} + 0.982550E+00 Z^{-2}}$$

BINARY REPRESENTATION OF NUMERATOR COEFFICIENTS

R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13

N2(2,3) = 1.000000

1 0 0 0 0 0 0 0 0 0 0 0 0 0

ABSOLUTE BINARY EQUIVALENT = 1.000000
(THIS IS 1.000000 OF THE ACTUAL VALUE)

N2(2,2) = -1.749900

1 1 1 1 0 0 0 0 0 1 0 1 1 1

ABSOLUTE BINARY EQUIVALENT = 1.877805
(THIS IS 0.999949 OF THE ACTUAL VALUE)

N2(2,1) = 1.001700

0 1 1 1 1 1 1 1 1 1 0 0 0 1

ABSOLUTE BINARY EQUIVALENT = 0.998169
(THIS IS 0.999999 OF THE ACTUAL VALUE)

BINARY REPRESENTATION OF DENOMINATOR COEFFICIENTS

R00 R01 R02 R03 R04 R05 R06 R07 R08 R09 R10 R11 R12 R13

D2(2,3) = 1.000000

1 0 0 0 0 0 0 0 0 0 0 0 0 0

ABSOLUTE BINARY EQUIVALENT = 1.000000
(THIS IS 1.000000 OF THE ACTUAL VALUE)

D2(2,2) = -1.809500

1 1 1 0 0 1 1 1 1 0 0 1 1 1

ABSOLUTE BINARY EQUIVALENT = 1.809446
(THIS IS 0.999970 OF THE ACTUAL VALUE)

D2(2,1) = 0.982550

0 1 1 1 1 1 0 1 1 1 0 0 0 1

ABSOLUTE BINARY EQUIVALENT = 0.982544
(THIS IS 0.999994 OF THE ACTUAL VALUE)

```

C*****FIR00010
C                                          *FIR00020
C          APPENDIX J                      *FIR00030
C                                          *FIR00040
C          FORTRAN PROGRAM FIR4           *FIR00050
C          ADAPTIVE TRANSVERSAL FILTER    *FIR00060
C                                          *FIR00070
C THIS PROGRAM WILL OBTAIN THE OPTIMAL FILTER WEIGHTS FOR THE FIR *FIR00080
C FILTER OF ORDER FOUR. THE ALGORITHM BEGINS WITH A TRIAL REGION *FIR00090
C (-40,40) FOR EACH OF THE WEIGHTS AND DOES A SUCCESSIVE ITERATION *FIR00100
C OF THE ERROR FUNCTION WHILE TRANSFORMING THE WEIGHTS TO OPTIMAL *FIR00110
C VALUES. WHEN THE WEIGHTS CONVERGE TO OPTIMAL VALUES THEN THE *FIR00120
C ITERATION STOPS. THEN WE MAY TEST THE ADEQUACY OF THE WEIGHTS *FIR00130
C SO OBTAINED THROUGH SUBSEQUENT SIMULATION IN THE FORTRAN PROGRAM *FIR00140
C FIR4SIM WHICH FOLLOWS AS APPENDIX K.   *FIR00150
C                                          *FIR00160
C*****FIR00170
C      REAL WS(4),WU(4),WL(4),R,JE      FIR00180
C      INTEGER NTA,NPR,NAV,NV,IP        FIR00190
C WS(1) IS THE STARTING GUESS          FIR00200
C      WS(1)=.1                         FIR00210
C      WS(2)=1.                         FIR00220
C      WS(3)=1.                         FIR00230
C      WS(4)=1.                         FIR00240
C WL(1) IS THE LOWER LIMIT FOR THE I'TH VARIABLE FIR00250
C WU(1) IS THE UPPER LIMIT FOR THE I'TH VARIABLE FIR00260
C      WL(1)=-14.                      FIR00270
C      WU(1)=14.                       FIR00280
C      WL(2)=-14.                      FIR00290
C      WU(2)=14.                       FIR00300
C      WL(3)=-14.                      FIR00310
C      WU(3)=14.                       FIR00320
C      WL(4)=-14.                      FIR00330
C      WU(4)=14.                       FIR00340
C A DESCRIPTION OF THE FOLLOWING PARAMETERS IS DISCUSSED IN BOXPLX FIR00350
C      R=.13.                          FIR00360
C      NTA=1400                        FIR00370
C      NPR=100                        FIR00380
C      NAV=0                          FIR00390
C      NV=4                           FIR00400
C      IP=0                           FIR00410
C PERFORM ITERATION ROUTINE FOR WEIGHT OPTIMIZATION FIR00420
C CALL BOXPLX(NV,NAV,NPR,NTA,R,WS,IP,WU,WL,VMN,IER) FIR00430
C WRITE (6,25)                        FIR00440
25  FORMAT(1X,' OPTIMAL GAINS',/)      FIR00450
C DO 30 I=1,4                         FIR00460
30  WRITE(6,40)I,WS(I)                FIR00470
40  FORMAT(1X,'W(',I2,')=',F14.7)    FIR00480
C STOP                                FIR00490
C END                                  FIR00500
C*****FIR00510
C      SUBROUTINE FIR(XX)               FIR00520
C SUBROUTINE FIR(XX) SIMULATES THE FIR FILTER FIR00530
C COMMON J                             FIR00540
C      REAL*8 J,W0,W1,W2,W3,X1,X2,X3,INPUT,OUTPUT FIR00550

```

.FILE: FIR4P FORTRAN A1

DIMENSION XX(4), DESIRE(105)	FIR00560
C INITIAL CONDITIONS	FIR00570
ETIME=100.	FIR00580
T=0.0	FIR00590
ICOUNT=2	FIR00600
C INITIALIZE THE COST (CUMULATIVE ERROR) FUNCTION	FIR00610
J=0.0	FIR00620
C GAIN COEFFICIENTS TO BE OPTIMIZED	FIR00630
W0=XX(1)	FIR00640
W1=XX(2)	FIR00650
W2=XX(3)	FIR00660
W3=XX(4)	FIR00670
C SHIFT REGISTERS	FIR00680
X1=0.0	FIR00690
X2=0.0	FIR00700
X3=0.0	FIR00710
C SIMULATE DESIRED OUTPUT SIGNAL	FIR00720
DO 15 I=1,105	FIR00730
15 DESIRE(I)=-1.0	FIR00740
DO 16 I=1,11	FIR00750
16 DESIRE(I+44)=1.0	FIR00760
C	FIR00770
K=1	FIR00780
C OUTPUT HEADING	FIR00790
C WRITE(6,99)	FIR00800
C 99 FORMAT(' ', 'FIR TRANSVERSAL FILTER SIMULATION RESULTS', ///,	FIR00810
C &' TIME INPUT SIMULATED OUTPUT DESIRED OUTPUT', /)	FIR00820
C LOOP FOR 100 SAMPLE ITERATIONS	FIR00830
200 CONTINUE	FIR00840
C SIMULATED INPUT SIGNAL	FIR00850
INPUT=SIN(.1*T)*COS(.1*T)*(2.+COS(.1*T))	FIR00860
C INPUT=-.0004*T**2+.04*T	FIR00870
C SIMULATED OUTPUT SIGNAL FROM FIR FILTER	FIR00880
OUTPUT=W0*INPUT+W1*X1+W2*X2+W3*X3	FIR00890
C WHEN TO PRINTOUT	FIR00900
IF (ICOUNT.EQ. 2) GO TO 50	FIR00910
GO TO 300	FIR00920
C PRINTOUT	FIR00930
50 CONTINUE	FIR00940
C EASYPLOT OUTPUT OPTION	FIR00950
C WRITE (6,100) T, INPUT, OUTPUT, DESIRE(K)	FIR00960
C 100 FORMAT(2X, F8.4, 'X', F8.4, 'Y', F8.4, 'Z', F8.4)	FIR00970
C SCREEN OUTPUT OPTION	FIR00980
C WRITE (6,100) T, INPUT, OUTPUT, DESIRE(K)	FIR00990
C 100 FORMAT(1X, 'TIME=', F8.4, 'X=', F8.4, 'Y=', F8.4, 'Z=', F8.4, 'OUTPUT=', F8.4, 'X',	FIR01000
C &'DESIRED OUTPUT', F8.4)	FIR01010
ICOUNT=1	FIR01020
C TEST IF WANT TO STOP	FIR01030
300 IF (T.GE.ETIME) GO TO 400	FIR01040
C JE=ERROR FUNCTION	FIR01050
JE=(OUTPUT-DESIRE(K))**2	FIR01060
C J=COST FUNCTION (CUMULATIVE ERROR)	FIR01070
J=J+JE	FIR01080
C STEP SIZE DELT	FIR01090
DELT=1.0	FIR01100

```

      T=T+DELT
      K=K+1
      ICOUNT=ICOUNT+1
      X3=X2
      X2=X1
      X1=INPUT
      GO TO 200
C  OUTPUT OPTIMAL WEIGHTS
400  WRITE(6,500) JE,W0,W1,W2,W3
500  FORMAT(' ',1X,' J =',E15.9,2X,
1    'W0=',F15.7,2X,'W1=',F15.7,2X,'W2=',F15.7,2X,'W3=',F15.7)
      RETURN
      FND
C  .....
C
C  SUBROUTINE BOXPLX                      (CATEGORY H0)
C
C  PURPOSE
C
C  BOXPLX IS A SUBROUTINE USED TO SOLVE THE PROBLEM OF LOCATING
C  A MINIMUM (OR MAXIMUM) OF AN ARBITRARY OBJECTIVE FUNCTION
C  SUBJECT TO ARBITRARY EXPLICIT AND/OR IMPLICIT CONSTRAINTS BY
C  THE COMPLEX METHOD OF M.J. BOX. EXPLICIT CONSTRAINTS ARE
C  DEFINED AS UPPER AND LOWER BOUNDS ON THE INDEPENDENT VARIABLES.
C  IMPLICIT CONSTRAINTS MAY BE ARBITRARY FUNCTIONS OF THE VAR-
C  IABLES. TWO FUNCTION SUBPROGRAMS TO EVALUATE THE OBJECTIVE
C  FUNCTION AND IMPLICIT CONSTRAINTS, RESPECTIVELY, MUST BE
C  SUPPLIED BY THE USER (SEE EXAMPLE BELOW). BOXPLX ALSO HAS
C  THE OPTION TO PERFORM INTEGER PROGRAMMING, WHERE THE VALUES
C  OF THE INDEPENDENT VARIABLES ARE RESTRICTED TO INTEGERS.
C
C  USAGE
C
C  CALL BOXPLX (NV,NAV,NPR,NTA,R,XS,IP,XU,XL,YMN,IER)
C
C  DESCRIPTION OF PARAMETERS
C
C  NV    AN INTEGER INPUT DEFINING THE NUMBER OF INDEPENDENT
C  VARIABLES OF THE OBJECTIVE FUNCTION TO BE MINIMIZED.
C  NOTE: MAXIMUM NV + NAV IS PRESENTLY 50. MAXIMUM NV IS
C  25. IF THESE LIMITS MUST BE EXCEEDED, PUNCH A SOURCE
C  DECK IN THE USUAL MANNER, AND CHANGE THE DIMENSION
C  STATEMENTS.
C
C  NAV   AN INTEGER INPUT DEFINING THE NUMBER OF AUXILIARY VAR-
C  IABLES THE USER WISHES TO DEFINE FOR HIS OWN CONVENIENCE.
C  TYPICALLY HE MAY WISH TO DEFINE THE VALUE OF EACH IMPLICIT
C  CONSTRAINT FUNCTION AS AN AUXILIARY VARIABLE. IF THIS
C  IS DONE, THE OPTIONAL OUTPUT FEATURE OF BOXPLX CAN BE
C  USED TO OBSERVE THE VALUES OF THOSE CONSTRAINTS AS THE
C  SOLUTION PROGRESSES. AUXILIARY VARIABLES, IF USED,
C  SHOULD BE EVALUATED IN FUNCTION KE (DEFINED BELOW).
C  NAV MAY BE ZERO.

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FIR01110
FIR01120
FIR01130
FIR01140
FIR01150
FIR01160
FIR01170
FIR01180
FIR01190
FIR01200
FIR01210
FIR01220
FIR01230
FIR01240
FIR01250
FIR01260
FIR01270
FIR01280
FIR01290
FIR01300
FIR01310
FIR01320
FIR01330
FIR01340
FIR01350
FIR01360
FIR01370
FIR01380
FIR01390
FIR01400
FIR01410
FIR01420
FIR01430
FIR01440
FIR01450
FIR01460
FIR01470
FIR01480
FIR01490
FIR01500
FIR01510
FIR01520
FIR01530
FIR01540
FIR01550
FIR01560
FIR01570
FIR01580
FIR01590
FIR01600
FIR01610
FIR01620
FIR01630
FIR01640
FIR01650

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C      NPR  INPUT INTEGER CONTROLLING THE FREQUENCY OF OUTPUT DESIRED  FIR01660
C      FOR DIAGNOSTIC PURPOSES.  IF NPR .LE. 0, NO OUTPUT WILL BE  FIR01670
C      PRODUCED BY BOXPLX.  OTHERWISE, THE CURRENT COMPLEX OF  FIR01680
C      K= 2*NV VERTICES AND THEIR CENTROID WILL BE OUTPUT AFTER  FIR01690
C      EACH NPR PERMISSIBLE TRIALS.  THE NUMBER OF TOTAL TRIALS,  FIR01700
C      NUMBER OF FEASIBLE TRIALS, NUMBER OF FUNCTION EVALUATIONS  FIR01710
C      AND NUMBER OF IMPLICIT CONSTRAINT EVALUATIONS ARE IN-  FIR01720
C      CLUDED IN THE OUTPUT.  FIR01730
C      ADDITIONALLY, (WHEN NPR .GT. 0) THE SAME INFORMATION  FIR01740
C      WILL BE OUTPUT:  FIR01750
C      FIR01760
C      1) IF THE INITIAL POINT IS NOT FEASIBLE,  FIR01770
C      2) AFTER THE FIRST COMPLETE COMPLEX IS GENERATED,  FIR01780
C      3) IF A FEASIBLE VERTEX CANNOT BE FOUND AT SOME TRIAL,  FIR01790
C      4) IF THE OBJECTIVE VALUE OF A VERTEX CANNOT BE MADE  FIR01800
C      NO-LONGER-WORST.  FIR01810
C      5) IF THE LIMIT ON TRIALS (NTA) IS REACHED AND,  FIR01820
C      6) WHEN THE OBJECTIVE FUNCTION HAS BEEN UNCHANGED FOR  FIR01830
C      2*NV TRIALS, INDICATING A LOCAL MINIMUM HAS BEEN  FIR01840
C      FOUND.  FIR01850
C      FIR01860
C      IF THE USER WISHES TO TRACE THE PROGRESS OF A SOLUTION,  FIR01870
C      A CHOICE OF NPR = 25, 50 OR 100 IS RECOMMENDED.  FIR01880
C      FIR01890
C      NTA  INTEGER INPUT OF LIMIT ON THE NUMBER OF TRIALS ALLOWED  FIR01900
C      IN THE CALCULATION.  IF THE USER INPUTS NTA .LE. 0, A  FIR01910
C      DEFAULT VALUE OF 2000 IS USED.  WHEN THIS LIMIT IS REACHED  FIR01920
C      CONTROL RETURNS TO THE CALLING PROGRAM WITH THE BEST  FIR01930
C      ATTAINED OBJECTIVE FUNCTION VALUE IN YMN, AND THE BEST  FIR01940
C      ATTAINED SOLUTION POINT IN XS.  FIR01950
C      FIR01960
C      R    A REAL NUMBER INPUT TO DEFINE THE FIRST RANDOM NUMBER  FIR01970
C      USED IN DEVELOPING THE INITIAL COMPLEX OF 2*NV VERTICES.  FIR01980
C      (0. .GT. R .LT. 1.) IF R IS NOT WITHIN THESE BOUNDS,  FIR01990
C      IT WILL BE REPLACED BY 1./3. .  FIR02000
C      FIR02010
C      XS   INPUT REAL ARRAY DIMENSIONED AT LEAST NV+NAV. THE FIRST  FIR02020
C      NV MUST CONTAIN A FEASIBLE ORIGIN FOR STARTING THE CAL-  FIR02030
C      CULATION.  THE LAST NAV NEED NOT BE INITIALIZED.  UPON  FIR02040
C      RETURN FROM BOXPLX, THE FIRST NV ELEMENTS OF THE ARRAY  FIR02050
C      CONTAIN THE COORDINATES OF THE MINIMUM OBJECTIVE FUNCTION,  FIR02060
C      AND THE REMAINING NAV (NAV .GE. 0) CONTAIN THE VALUES OF  FIR02070
C      THE CORRESPONDING AUXILIARY VARIABLES.  FIR02080
C      FIR02090
C      IP   INTEGER INPUT FOR OPTIONAL INTEGER PROGRAMMING.  IF IP=1,  FIR02100
C      THE VALUES OF THE INDEPENDENT VARIABLES WILL BE REPLACED  FIR02110
C      WITH INTEGER VALUES (STILL STORED AS REAL*4).  FIR02120
C      FIR02130
C      XU   A REAL ARRAY DIMENSIONED AT LEAST NV INPUTTING THE UPPER  FIR02140
C      BOUND ON EACH INDEPENDENT VARIABLE, (EACH EXPLICIT CON-  FIR02150
C      STRAINT).  INPUT VALUES ARE SLIGHTLY ALTERED BY BOXPLX.  FIR02160
C      FIR02170
C      XL   A REAL ARRAY DIMENSIONED AT LEAST NV INPUTTING THE LOWER  FIR02180
C      BOUND ON EACH INDEPENDENT VARIABLE, (EACH EXPLICIT CON-  FIR02190
C      STRAINT).  NOTE:  FOR BOTH XU AND XL CHOOSE REASONABLE  FIR02200

```

FILE: FIR4SIMP FORTRAN A1

```
      K=1
C  OUTPUT HEADING
      WRITE(6,99)
      99 FORMAT(' ', 'FIR TRANSVERSAL FILTER SIMULATION RESULTS', ///,
      &' TIME INPUT SIMULATED OUTPUT DESIRED OUTPUT', /)
C  LOOP FOR 100 SAMPLE ITERATIONS
      200 CONTINUE
C  SIMULATED INPUT SIGNAL (600 HZ + 1200 HZ + 1800 HZ)
      INPUT=SIN(.1*T)*COS(.1*T)*(2.+COS(.1*T))
C  SIMULATED OUTPUT SIGNAL FROM FIR FILTER
      OUTPUT=WO*INPUT+W1*X1+W2*X2+W3*X3
C  WHEN TO PRINTOUT
      IF (ICOUNT.EQ. 2) GO TO 50
      GO TO 300
C  PRINTOUT
      50 CONTINUE
C  EASYPLOT OUTPUT OPTION
      WRITE (6,100) T, INPUT, OUTPUT, DESIRE(K)
      100 FORMAT(2X, F8.4, 1X, F8.4, 3X, F8.4, 13X, F8.4)
C  SCREEN OUTPUT OPTION
      WRITE (6,100) T, INPUT, OUTPUT, DESIRE(K)
C  100 FORMAT(1X, 'TIME=', F7.3, 5X, 'INPUT=', F8.4, 5X, 'OUTPUT=', F8.4, 5X,
C  &'DESIRED OUTPUT=', F8.4)
      ICOUNT=1
C  TEST IF WANT TO STOP
      300 IF (T.GE.ETIME) GO TO 400
C  JE=ERROR FUNCTION
      JE=(OUTPUT-DESIRED(K))**2
C  J=COST FUNCTION (CUMULATIVE ERROR)
      J=J+JE
C  STEP SIZE DELT
      DELT=1.0
      T=T+DELT
      K=K+1
      ICOUNT=ICOUNT+1
      X3=X2
      X2=X1
      X1=INPUT
      GO TO 200
400 RETURN
      END
```

FIR00560
FIR00570
FIR00580
FIR00590
FIR00600
FIR00610
FIR00620
FIR00630
FIR00640
FIR00650
FIR00660
FIR00670
FIR00680
FIR00690
FIR00700
FIR00710
FIR00720
FIR00730
FIR00740
FIR00750
FIR00760
FIR00770
FIR00780
FIR00790
FIR00800
FIR00810
FIR00820
FIR00830
FIR00840
FIR00850
FIR00860
FIR00870
FIR00880
FIR00890
FIR00900
FIR00910
FIR00920
FIR00930
FIR00940
FIR00950
FIR00960

```

C*****FIR00010
C                                     *FIR00020
C                                     *FIR00030
C                                     *FIR00040
C                                     *FIR00050
C                                     *FIR00060
C                                     *FIR00070
C                                     *FIR00080
C                                     *FIR00090
C                                     *FIR00100
C                                     *FIR00110
C                                     *FIR00120
C                                     *FIR00130
C*****FIR00140
C      REAL W(4),R,JE
C      W(1) IS THE CALCULATED OPTIMAL GAIN
C      W(1)=-7.5060358
C      W(2)=7.5403662
C      W(3)=4.7097464
C      W(4)=-5.3987589
C      WRITE (6,25)
25  FORMAT(1X,' OPTIMAL GAINS',/)
C      DO 30 I=1,4
C      WRITE(6,40)I,W(I)
40  FORMAT(1X,'W(',I2,')=' ,F14.7)
C      CALL FIR(W)
C      STOP
C      END
C*****FIR00290
C      SUBROUTINE FIR(XX)
C      SUBROUTINE FIR(XX) SIMULATES THE FIR ADAPTIVE TRANSVERSAL FILTER
C      COMMON J
C      REAL*8 J,W0,W1,W2,W3,X1,X2,X3,INPUT,OUTPUT
C      DIMENSION XX(4),DESIRE(105)
C      INITIAL CONDITIONS
C      ETIME=100.
C      T=0.0
C      ICOUNT=2
C      INITIALIZE THE COST (CUMULATIVE ERROR) FUNCTION
C      J=0.0
C      GAIN COEFFICIENTS TO BE OPTIMIZED
C      W0=XX(1)
C      W1=XX(2)
C      W2=XX(3)
C      W3=XX(4)
C      SHIFT REGISTERS
C      X1=0.0
C      X2=0.0
C      X3=0.0
C      SIMULATE DESIRED OUTPUT SIGNAL
C      DO 15 I=1,105
15  DESIRE(I)=-1.0
C      DO 16 I=1,11
16  DESIRE(I+44)=1.0
C

```


FILE: FIR4P FORTRAN A1

KE=0
RETURN
END

FIR08260
FIR08270
FIR08280

FILE: FIR4P FORTRAN A1

```
1 ' NEW MIN IS ',E15.7)
56 FORMAT ('OMIN OBJECTIVE FUNCTION IS ',E15.7)
END
SUBROUTINE FBV (K,FUN,M)
DIMENSION FUN(50)
M = 1
C
DO 1 I=2,K
IF (FUN(M).LE.FUN(I)) GO TO 1
M = I
1 CONTINUE
C
RETURN
END
SUBROUTINE BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FN,C,IK)
DIMENSION V(50,50), FN(50), C(25)
WRITE (6,4) NT,NPT,NFE,NCE
C
DO 1 I=1,K
WRITE (6,5) FN(I),(V(J,I),J=1,NV)
IF (NVT.LE.NV) GO TO 1
NVP = NV+1
WRITE (6,6) (V(J,I),J=NVP,NVT)
1 CONTINUE
C
IF (IK.NE.0) GO TO 2
C
WRITE (6,7) (C(I),I=1,NV)
RETURN
2 IF (IK.GE.0) GO TO 3
WRITE (6,8) (C(I),I=1,NV)
RETURN
3 WRITE (6,9) IK,(C(I),I=1,NV)
RETURN
C
4 FORMAT ('ONO. TOTAL TRIALS = ',15,4X,'NO. FEASIBLE TRIALS = ',
115,4X,'NO. FUNCTION EVALUATIONS = ',15,4X,'NO. CONSTRAINT EVALUATIONS = ',15,4X,'NO. INDEPENDENT VARIABLES/DEPENDENT OR IMPLICIT CONSTRAINTS')
5 FORMAT (1H ,E18.7,2X,7E14.7/(21X,7E14.7))
6 FORMAT (21X,7E14.7)
7 FORMAT (10HOCENTROID 11X,7E14.7/(21X,7E14.7))
8 FORMAT ('O BEST VERTEX',7X,7E14.7/(21X,7E14.7))
9 FORMAT ('OCENTROID LESS VX',12,2X,7E14.7/(21X,7E14.7))
END
FUNCTION FE(X)
DIMENSION X(4)
REAL J
COMMON J
CALL FIR(X)
FE=J
RETURN
END
FUNCTION KE(X)
DIMENSION X(4)
```

FIR07710
FIR07720
FIR07730
FIR07740
FIR07750
FIR07760
FIR07770
FIR07780
FIR07790
FIR07800
FIR07810
FIR07820
FIR07830
FIR07840
FIR07850
FIR07860
FIR07870
FIR07880
FIR07890
FIR07900
FIR07910
FIR07920
FIR07930
FIR07940
FIR07950
FIR07960
FIR07970
FIR07980
FIR07990
FIR08000
FIR08010
FIR08020
FIR08030
FIR08040
FIR08050
FIR08060
FIR08070
FIR08080
FIR08090
FIR08100
FIR08110
FIR08120
FIR08130
FIR08140
FIR08150
FIR08160
FIR08170
FIR08180
FIR08190
FIR08200
FIR08210
FIR08220
FIR08230
FIR08240
FIR08250

FILE: FIR4P FORTRAN A1

C	IF NOT, GO TO NEW TRIAL.		FIR07160
	40 IF (NT.GE.NTA) GO TO 41		FIR07170
C			FIR07180
C	NEXT-TO-WORST VERTEX NOW BECOMES WORST.		FIR07190
	J = JN		FIR07200
	GO TO 17		FIR07210
	41 IER = 3		FIR07220
	IF (NPR.GT.0) WRITE (6,54)		FIR07230
C			FIR07240
C	COLLECTOR POINT FOR ALL ENDINGS.		FIR07250
C	1) CANNOT DEVELOP FEASIBLE VERTEX.	IER = 1	FIR07260
C	2) CANNOT DEVELOP A NO-LONGER-WORST VERTEX.	IER = 2	FIR07270
C	3) FUNCTION VALUE UNCHANGED FOR K TRIALS.	IER = 0	FIR07280
C	4) LIMIT ON TRIALS REACHED.	IER = 3	FIR07290
C	5) CANNOT FIND FEASIBLE VERTEX AT START.	IER = -1	FIR07300
	42 CONTINUE		FIR07310
C			FIR07320
C	FIND BEST VERTEX.		FIR07330
	CALL FBV (K,FUN,M)		FIR07340
	IF (IER.GE.3) GO TO 44		FIR07350
C			FIR07360
C	RESTART IF THIS SOLUTION IS SIGNIFICANTLY BETTER THAN THE PREVIOUS,		FIR07370
C	OR IF THIS IS THE FIRST TRY.		FIR07380
	IF (NPR.LE.0) GO TO 43		FIR07390
	WRITE (6,55) (M,YMN,FUN(M))		FIR07400
	43 IF (FUN(M).GE.YMN) GO TO 47		FIR07410
	IF (ABS(FUN(M)-YMN).LE.AMAX1(EP,EP*YMN)) GO TO 47		FIR07420
C			FIR07430
C	GIVE IT ANOTHER TRY UNLESS LIMIT ON TRIALS REACHED.		FIR07440
	44 YMN = FUN(M)		FIR07450
	FUN(1) = FUN(M)		FIR07460
C			FIR07470
	DO 45 I=1,NV		FIR07480
	CEN(1) = V(1,M)		FIR07490
	SUM(1) = V(1,M)		FIR07500
	45 V(1,1) = V(1,M)		FIR07510
C			FIR07520
	DO 46 I=1,NVT		FIR07530
	46 XS(1) = V(1,M)		FIR07540
C			FIR07550
	IF (IER.LT.3) GO TO 6		FIR07560
	47 IF (NPR.LE.0) GO TO 48		FIR07570
	CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,V(1,M),-1)		FIR07580
	WRITE (6,56) FUN(M)		FIR07590
	48 RETURN		FIR07600
C			FIR07610
	49 FORMAT (50H0INDEX AND DIRECTION OF OUTLYING VARIABLE AT START15)		FIR07620
	50 FORMAT (50H0IMPLICIT CONSTRAINT VIOLATED AT START. DEAD END.)		FIR07630
	51 FORMAT ('0CANNOT FIND FEASIBLE',14,'TH VERTEX OR CENTROID AT START		FIR07640
	1.')		FIR07650
	52 FORMAT (10H0AT TRIAL 14,54H CANNOT FIND FEASIBLE VERTEX WHICH IS N		FIR07660
	10 LONGER WORST,14,15X,'RESTART FROM BEST VERTEX.')		FIR07670
	53 FORMAT (40H0FUNCTION HAS BEEN ALMOST UNCHANGED FOR 15,7H TRIALS)		FIR07680
	54 FORMAT (27H0LIMIT ON TRIALS EXCEEDED.)		FIR07690
	55 FORMAT ('0BEST VERTEX IS NO.',13,' OLD MIN WAS ',E15.7,		FIR07700

FILE: FIR4P FORTRAN A1

```

      IF (IP.EQ.1) VT = AINT(VT+.5)
29  V(I,J) = AMAX1(AMIN1(VT,BU(I)),BL(I))
C
      GO TO 32
C
30  DO 31 I=1,NV
      VT = .5*(CEN(I)+V(I,J))
      IF (IP.EQ.1) VT = AINT(VT+.5)
      V(I,J) = VT
31  CONTINUE
C
32  IF (LIMT.LT.NLIM) GO TO 33
C
C   CANNOT MAKE THE 'J'TH VERTEX NO LONGER WORST BY DISPLACING TOWARD
C   THE CENTROID OR BY OVER-REFLECTING THRU THE BEST VERTEX.
      IER = 2
      IF (NPR.LE. 0) GO TO 42
      WRITE (6,52) NT, J
      CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,CEN,J)
      GO TO 42
33  NT = NT+1
      GO TO 20
C
C   SUCCESS: WE HAVE A REPLACEMENT FOR VERTEX J.
34  FUN(J) = FUNTRY
      FUNOLD = FUNTRY
      NPT = NPT+1
C
C   EVERY 100'TH PERMISSIBLE TRIAL, RECOMPUTE CENTROID SUMMATION TO
C   AVOID CREEPING ERROR.
      IF (MOD(NPT,100).NE.0) GO TO 37
C
      DO 36 I=1,NV
      SUM(I) = 0.
C
      DO 35 N=1,K
35  SUM(I) = SUM(I)+V(I,N)
C
      CEN(I) = SUM(I)/FK
36  CONTINUE
C
      LC = 0
      GO TO 39
C
37  DO 38 I=1,NV
38  SUM(I) = SUM(I)+V(I,J)
C
      LC = J
C
39  IF (NPR.LE.0) GO TO 40
      IF (MOD(NPT,NPR).NE.0) GO TO 40
C
      CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,CEN,LC)
C
C   HAS THE MAX. NUMBER OF TRIALS BEEN REACHED WITHOUT CONVERGENCE?

```

FIR06610
FIR06620
FIR06630
FIR06640
FIR06650
FIR06660
FIR06670
FIR06680
FIR06690
FIR06700
FIR06710
FIR06720
FIR06730
FIR06740
FIR06750
FIR06760
FIR06770
FIR06780
FIR06790
FIR06800
FIR06810
FIR06820
FIR06830
FIR06840
FIR06850
FIR06860
FIR06870
FIR06880
FIR06890
FIR06900
FIR06910
FIR06920
FIR06930
FIR06940
FIR06950
FIR06960
FIR06970
FIR06980
FIR06990
FIR07000
FIR07010
FIR07020
FIR07030
FIR07040
FIR07050
FIR07060
FIR07070
FIR07080
FIR07090
FIR07100
FIR07110
FIR07120
FIR07130
FIR07140
FIR07150

GO TO 24	FIR06060
C	FIR06070
C CONSTRAINT VIOLATION: MOVE NEW POINT TOWARD CENTROID.	FIR06080
C	FIR06090
22 DO 23 I=1,NV	FIR06100
VT = .5*(CEN(I)+V(I,J))	FIR06110
IF (IP.EQ.1) VT = AINT(VT+.5)	FIR06120
V(I,J) = VT	FIR06130
23 CONTINUE	FIR06140
C	FIR06150
24 NT = NT+1	FIR06160
25 CONTINUE	FIR06170
C	FIR06180
IER = 1	FIR06190
C	FIR06200
C CANNOT GET FEASIBLE VERTEX BY MOVING TOWARD CENTROID,	FIR06210
C OR BY OVER-REFLECTING THRU THE BEST VERTEX.	FIR06220
IF (NPR.LE.0) GO TO 42	FIR06230
WRITE (6,52) NT,J	FIR06240
CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,CEN,J)	FIR06250
GO TO 42	FIR06260
C	FIR06270
C FEASIBLE VERTEX FOUND, EVALUATE THE OBJECTIVE FUNCTION.	FIR06280
26 NFE = NFE+1	FIR06290
FUNTRY = FE(V(1,J))	FIR06300
C	FIR06310
C TEST TO SEE IF FUNCTION VALUE HAS NOT CHANGED.	FIR06320
AFO = ABS(FUNTRY-FUNOLD)	FIR06330
AMX = AMAX1(ABS(EP*FUNOLD),EP)	FIR06340
C	FIR06350
C ACTIVATE THE FOLLOWING TWO STATEMENTS FOR DIAGNOSTIC PURPOSES ONLY.	FIR06360
C WRITE (6,99) J,AFO,AMX,FUNTRY,FUNOLD,FUN(J),FUN(JN),NTFS,N	FIR06370
C 99 FORMAT (1X,13,6E15.7,2I5)	FIR06380
C IF (AFO.GT.AMX) GO TO 27	FIR06390
NTFS = NTFS+1	FIR06400
IF (NTFS.LT.NCT) GO TO 28	FIR06410
IER = 0	FIR06420
IF (NPR.LE.0) GO TO 42	FIR06430
WRITE (6,53) K	FIR06440
CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,CEN,0)	FIR06450
GO TO 42	FIR06460
27 NTFS = 0	FIR06470
C	FIR06480
C IS THE NEW VERTEX NO LONGER WORST?	FIR06490
28 IF (FUNTRY.LT.FUN(JN)) GO TO 34	FIR06500
C	FIR06510
C TRIAL VERTEX IS STILL WORST; ADJUST TOWARD CENTROID.	FIR06520
C EVERY 'KV'TH TIME, OVER-REFLECT THE OFFENDING VERTEX THROUGH THE	FIR06530
C BEST VERTEX.	FIR06540
LIMT = LIMT+1	FIR06550
IF (MOD(LIMT,KV).NE.0) GO TO 30	FIR06560
CALL FBV (K,FUN,M)	FIR06570
C	FIR06580
DO 29 I=1,NV	FIR06590
VT = BETA*V(I,M)-ALPHA*V(I,J)	FIR06600

C		FIR05510
C	FIND THE WORST VERTEX, THE 'J'TH.	FIR05520
	J = 1	FIR05530
C		FIR05540
	DO 16 I=2,K	FIR05550
	IF (FUN(J).GE.FUN(I)) GO TO 16	FIR05560
	J = I	FIR05570
	16 CONTINUE	FIR05580
C		FIR05590
C	BASIC LOOP. ELIMINATE EACH WORST VERTEX IN TURN. IT MUST BECOME	FIR05600
C	NO LONGER WORST, NOT MERELY IMPROVED. FIND NEXT-TO-WORST VERTEX,	FIR05610
C	THE 'JN'TH ONE.	FIR05620
	17 JN = 1	FIR05630
	IF (J.EQ.1) JN = 2	FIR05640
C		FIR05650
	DO 18 I=1,K	FIR05660
	IF (I.EQ.J) GO TO 18	FIR05670
	IF (FUN(JN).GE.FUN(I)) GO TO 18	FIR05680
	JN = I	FIR05690
	18 CONTINUE	FIR05700
C		FIR05710
C	LIMIT = NUMBER OF MOVES DURING THIS TRIAL TOWARD THE CENTROID	FIR05720
C	DUE TO FUNCTION VALUE.	FIR05730
	LIMIT = 1	FIR05740
C		FIR05750
C	COMPUTE CENTROID AND OVER REFLECT WORST VERTEX.	FIR05760
C		FIR05770
	DO 19 I=1,NV	FIR05780
	VT = V(I,J)	FIR05790
	SUM(I) = SUM(I)-VT	FIR05800
	CEN(I) = SUM(I)/FKM	FIR05810
	VT = BETA*CEN(I)-ALPHA*VT	FIR05820
	IF (IP.EQ.1) VT = AINT(VT+.5)	FIR05830
C		FIR05840
C	INSURE THE EXPLICIT CONSTRAINTS ARE OBSERVED.	FIR05850
	19 V(I,J) = AMAX1(AMIN1(VT,BU(I)),BL(I))	FIR05860
C		FIR05870
	NT = NT+1	FIR05880
C		FIR05890
C	CHECK FOR IMPLICIT CONSTRAINT VIOLATION.	FIR05900
C		FIR05910
	20 DO 25 N=1,NLIM	FIR05920
	NCE = NCE+1	FIR05930
	IF (KE(V(1,J)).EQ.0) GO TO 26	FIR05940
C		FIR05950
C	EVERY 'KV'TH TIME, OVER-REFLECT THE OFFENDING VERTEX THROUGH THE	FIR05960
C	BEST VERTEX.	FIR05970
	IF (MOD(N,KV).NE.0) GO TO 22	FIR05980
	CALL FBV (K,FUN,M)	FIR05990
C		FIR06000
	DO 21 I=1,NV	FIR06010
	VT = BETA*V(I,M)-ALPHA*V(I,J)	FIR06020
	IF (IP.EQ.1) VT = AINT(VT+.5)	FIR06030
	21 V(I,J) = AMAX1(AMIN1(VT,BU(I)),BL(I))	FIR06040
C		FIR06050

FILE: FIR4P FORTRAN A1

	FUNOLD = FUN(1)	FIR04960
C	DO 15 I=2,K	FIR04970
	FI = FI+1.	FIR04980
	LIMT = 0	FIR04990
	7 LIMT = LIMT+1	FIR05000
C	END CALCULATION IF FEASIBLE CENTROID CANNOT BE FOUND.	FIR05010
C	IF (LIMT.GE.NLIM) GO TO 11	FIR05020
C	DO 8 J=1,NV	FIR05030
C	RANDOM NUMBER GENERATOR (RANDU)	FIR05040
C	IQR = IQR*65539	FIR05050
	IF (IQR.LT.0) IQR = IQR+2147483647+1	FIR05060
	RQX = IQR	FIR05070
	RQX = RQX*.4656613E-9	FIR05080
	V(J,1) = BL(J)+RQX*(BU(J)-BL(J))	FIR05090
	IF (IP.EQ.1) V(J,1)=AINT(V(J,1)+.5)	FIR05100
	8 CONTINUE	FIR05110
C	DO 10 L=1,NLIM	FIR05120
	NCE = NCE+1	FIR05130
	IF (KE(V(1,1)).EQ.0) GO TO 13	FIR05140
C	DO 9 J=1,NV	FIR05150
	VT = .5*(V(J,1)+CEN(J))	FIR05160
	IF (IP.EQ.1) VT = AINT(VT+.5)	FIR05170
	V(J,1) = VT	FIR05180
	9 CONTINUE	FIR05190
C	10 CONTINUE	FIR05200
C	11 IF (NPR.LE.0) GO TO 12	FIR05210
	WRITE (6,51) I	FIR05220
	CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,I,FUN,CEN,I)	FIR05230
	12 IER = -1	FIR05240
	GO TO 48	FIR05250
C	13 DO 14 J=1,NV	FIR05260
	SUM(J) = SUM(J)+V(J,1)	FIR05270
	14 CEN(J) = SUM(J)/FI	FIR05280
C	TRY TO ASSURE FEASIBLE CENTROID FOR STARTING.	FIR05290
C	NCE = NCE+1	FIR05300
	IF (KE(CEN).EQ.0) GO TO 60	FIR05310
	SUM(J) = SUM(J) -V(J,1)	FIR05320
	GO TO 7	FIR05330
	60 NFE = NFE+1	FIR05340
	FUN(1) = FE(V(1,1))	FIR05350
	15 CONTINUE	FIR05360
C	END OF LOOP SETTING OF INITIAL COMPLEX.	FIR05370
C	IF (NPR.LE.0) GO TO 17	FIR05380
	CALL BOUT (NT,NPT,NFE,NCE,NV,NVT,V,K,FUN,CEN,0)	FIR05390
		FIR05400
		FIR05410
		FIR05420
		FIR05430
		FIR05440
		FIR05450
		FIR05460
		FIR05470
		FIR05480
		FIR05490
		FIR05500

	NPT = 0	FIR04410
C	CURRENT NO. OF PERMISSIBLE TRIALS	FIR04420
	NTFS = 0	FIR04430
C	CURRENT NO. OF TIMES F HAS BEEN ALMOST UNCHANGED	FIR04440
C		FIR04450
C	CHECK FEASIBILITY OF START POINT	FIR04460
C		FIR04470
	DO 4 I=1,NV	FIR04480
	VT = XS(I)	FIR04490
	IF (BL(I).LE.VT) GO TO 1	FIR04500
	II = -1	FIR04510
	VT = BL(I)	FIR04520
	GO TO 2	FIR04530
	1 IF (BU(I).GE.VT) GO TO 3	FIR04540
	II = 1	FIR04550
	VT = BU(I)	FIR04560
	2 IF (NPR.GT.0) WRITE (6,49) II	FIR04570
	3 V(1,1) = VT	FIR04580
	CEN(I) = VT	FIR04590
	IF (IP.EQ.1) GO TO 4	FIR04600
	BL(I) = BL(I)+AMAX1(EP,EP*ABS(BL(I)))	FIR04610
	BU(I) = BU(I)-AMAX1(EP,EP*ABS(BU(I)))	FIR04620
	4 SUM(I) = VT	FIR04630
C		FIR04640
C		FIR04650
	NCE = 1	FIR04660
C	NUMBER OF CONSTRAINT EVALUATIONS	FIR04670
	I = 1	FIR04680
	IF (KE(V(1,1)).EQ.0) GO TO 5	FIR04690
	IF (NPR.LE.0) GO TO 12	FIR04700
	WRITE (6,50)	FIR04710
	GO TO 12	FIR04720
	5 NFE = 1	FIR04730
C		FIR04740
C	NUMBER OF VERTICES (K) = 2 TIMES NO. OF VARIABLES.	FIR04750
	K = 2*NV	FIR04760
C		FIR04770
C	NUMBER OF DISPLACEMENTS ALLOWED.	FIR04780
	NLIM = 5*NV+10	FIR04790
C		FIR04800
C	NUMBER OF CONSECUTIVE TRIALS WITH UNCHANGED FE TO TERMINATE.	FIR04810
	NCT = NLIM+NV	FIR04820
	ALPHA = 1.3	FIR04830
	FK = K	FIR04840
	FKM = FK-1.	FIR04850
	BETA = ALPHA+1.	FIR04860
C		FIR04870
C	INSURE SEED OF RANDOM NUMBER GENERATOR IS ODD.	FIR04880
	IQR = R*1.E7	FIR04890
	IF (MOD(IQR,2).EQ.0) IQR=IQR+101	FIR04900
C		FIR04910
C	SET UP INITIAL VERTICES	FIR04920
	FUN(1) = FE(V(1,1))	FIR04930
	YMN = FUN(1)	FIR04940
	6 FI = 1.	FIR04950


```

C      SUBROUTINE 'BOUT' AND FUNCTION 'FBV' ARE INTEGRAL PARTS OF      FIRO3860
C      THE BOXPLX PACKAGE.                                           FIRO3870
C                                                                    FIRO3880
C      TWO FUNCTIONS MUST BE SUPPLIED BY THE USER. THE FIRST, KE(X), FIRO3890
C      IS USED TO EVALUATE THE IMPLICIT CONSTRAINT. SET KE=0 AT THE FIRO3900
C      BEGINING OF THE FUNCTION, THEN EVALUATE THE IMPLICIT CON- FIRO3910
C      STRAINTS. IN THE EXAMPLE ABOVE, THE FIRST CONSTRAINT, X(3), FIRO3920
C      MUST BE WITHIN THE RANGE (0. .LE. X(3) .LE. 6.). THE SECOND FIRO3930
C      CONSTRAINT X(4), MUST BE .GE. 0. . IF EITHER CONSTRAINT IS FIRO3940
C      NOT WITHIN THESE BOUNDS, CONTROL IS TRANSFERRED TO STATEMENT 1, FIRO3950
C      AND KE IS SET TO "1" AND CONTROL IS RETURNED TO BOXPLX.      FIRO3960
C                                                                    FIRO3970
C      THE SECOND FUNCTION THE USER MUST PROVIDE EVALUATES THE OB- FIRO3980
C      JECTIVE FUNCTION. IT IS CALLED FE(X) AS SHOWN IN THE EXAM- FIRO3990
C      PLE ABOVE, AND FE MUST BE SET TO THE VALUE OF THE OBJECTIVE FIRO4000
C      FUNCTION CORRESPONDING TO CURRENT VALUES OF THE NV INDEPENDENT FIRO4010
C      VARIABLES IN ARRAY 'X'.                                       FIRO4020
C                                                                    FIRO4030
C      REFERENCES                                                    FIRO4040
C                                                                    FIRO4050
C      BOX, M. J., "A NEW METHOD OF CONSTRAINED OPTIMIZATION AND A FIRO4060
C      COMPARISON WITH OTHER METHODS", COMPUTER JOURNAL, 8 APR. '65, FIRO4070
C      PP. 45-52.                                                    FIRO4080
C                                                                    FIRO4090
C      BEVERIDGE G., AND SCHECHTER R., "OPTIMIZATION: THEORY AND FIRO4100
C      PRACTICE", MCGRAW-HILL, 1970.                                FIRO4110
C                                                                    FIRO4120
C      PROGRAMMER                                                  FIRO4130
C                                                                    FIRO4140
C      R.R. HILLEARY 1/1966.                                         FIRO4150
C      REVISED FOR SYSTEM 360 4/1967                                FIRO4160
C      CORRECTED 1/1969                                              FIRO4170
C      REVISED/EXTENDED BY L.NOLAN/R.HILLEARY 2/1975              FIRO4180
C      CORRECTED 8/1976                                              FIRO4190
C                                                                    FIRO4200
C      .....                                                       FIRO4210
C                                                                    FIRO4220
C      SUBROUTINE BOXPLX (NV,NAV,NPR,NTZ,RZ,XS,IP,BU,BL,YMN,IER)     FIRO4230
C                                                                    FIRO4240
C      DIMENSION V(50,50), FUN(50), SUM(25), CEN(25), XS(NV), BU(NV), BL( FIRO4250
C      1NV)                                                           FIRO4260
C                                                                    FIRO4270
C      KV = 5                                                         FIRO4280
C      EP = 1.E-6                                                     FIRO4290
C      NTA = 2000                                                      FIRO4300
C      IF (NTZ.GT.0) NTA = NTZ                                         FIRO4310
C      R = RZ                                                           FIRO4320
C      IF (R.LE.0..OR.R.GE.1.) R=1./3.                                FIRO4330
C      NVT = NV+NAV                                                    FIRO4340
C                                                                    FIRO4350
C                                                                    FIRO4360
C      TOTAL VARS, EXPLICIT PLUS IMPLICIT                             FIRO4370
C      NT = 0                                                         FIRO4380
C      CURRENT TRIAL NO.                                              FIRO4390
C                                                                    FIRO4400

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C      THE REPLACED VERTEX AND CENTROID OF ALL OTHER VERTICES.)      FIRO3310
C                                                                    FIRO3320
C      WHEN AN OVER-REFLECTION IS NOT FEASIBLE OR REMAINS WORST, IT  FIRO3330
C      IS CONSIDERED NOT-PERMISSIBLE AND IS DISPLACED HALFWAY TOWARD FIRO3340
C      THE CENTROID.  AFTER FOUR SUCH ATTEMPTS ARE MADE UNSUCCESSFULLY FIRO3350
C      EVERY FIFTH ATTEMPT IS MADE BY REFLECTING THE OFFENDING VERTEX FIRO3360
C      THROUGH THE PRESENT BEST VERTEX, INSTEAD OF THROUGH THE CEN-  FIRO3370
C      TROID.  IF 5*N+10 DISPLACEMENTS AND OVER-REFLECTIONS OCCUR  FIRO3380
C      WITHOUT A SUCCESSFUL (PERMISSIBLE) RESULT, THE CURRENT BEST  FIRO3390
C      VERTEX IS TAKEN AS AN INITIAL FEASIBLE POINT FOR A RESTART  FIRO3400
C      RUN OF THE COMPLETE PROCESS.  RESTARTING IS ALSO UNDERTAKEN  FIRO3410
C      WHEN 6*N+10 CONSECUTIVE TRIALS HAVE BEEN MADE WITH NO SIGNIF- FIRO3420
C      ICANT CHANGE IN THE VALUE OF THE OBJECTIVE FUNCTION.  IN ALL  FIRO3430
C      CASES, RESTARTING IS INHIBITED IF THE LAST RESTART DID NOT  FIRO3440
C      PRODUCE A SIGNIFICANT IMPROVEMENT IN THE MINIMUM ATTAINED.  FIRO3450
C                                                                    FIRO3460
C      IT IS RECOMMENDED THAT THE USER READ THE REFERENCE FOR  FIRO3470
C      FURTHER USEFUL INFORMATION.  IT SHOULD BE NOTED THAT THE  FIRO3480
C      ALGORITHM DEFINED THERE HAS BEEN ALTERED TO FIND THE  FIRO3490
C      CONSTRAINED MINIMUM, RATHER THAN THE MAXIMUM.  FIRO3500
C                                                                    FIRO3510
C                                                                    FIRO3520
C                                                                    FIRO3530
C      REMARKS  FIRO3540
C                                                                    FIRO3550
C      THE INTEGER PROGRAMMING OPTION WAS ADDED TO THIS PROGRAM  FIRO3560
C      AS SUGGESTED IN REFERENCE (2).  A MIXED INTEGER/CONTINUOUS  FIRO3570
C      VARIABLE VERSION OF BOXPLX WOULD BE EASY TO CREATE BY DE-  FIRO3580
C      CLARING "IP" TO BE AN ARRAY OF NV CONTROL VARIABLES WHERE IP  FIRO3590
C      (I)=1 WOULD INDICATE THAT THE I-TH VARIABLE IS TO BE CONFINED  FIRO3600
C      TO INTEGER VALUES.  EACH STATEMENT OF THE FORM 'IF (IP .EQ.  FIRO3610
C      1)' ETC. WOULD THEN NEED TO BE ALTERED TO 'IF (IP(I) .EQ. 1)'  FIRO3620
C      ETC., WHERE THE SUBSCRIPT IS APPROPRIATELY CHOSEN.  NORMALLY,  FIRO3630
C      XU AND XL VALUES ARE ALTERED TO BE AN EPSILON 'WITHIN' ACTUAL  FIRO3640
C      VALUES DECLARED BY THE USER.  THIS ADJUSTMENT IS NOT MADE  FIRO3650
C      WHEN IP=1.  FIRO3660
C                                                                    FIRO3670
C      NOTE:  NO NON-LINEAR PROGRAMMING ALGORITHM CAN GUARANTEE THAT  FIRO3680
C      THE ANSWER FOUND IS THE GLOBAL MINIMUM, RATHER THAN JUST A  FIRO3690
C      LOCAL MINIMUM.  HOWEVER, ACCORDING TO REF.2, THE COMPLEX  FIRO3700
C      METHOD HAS AN ADVANTAGE IN THAT IT TENDS TO FIND THE GLOBAL  FIRO3710
C      MINIMUM MORE FREQUENTLY THAN MANY OTHER NON-LINEAR PROGRAM-  FIRO3720
C      MING ALGORITHMS.  FIRO3730
C                                                                    FIRO3740
C      IT SHOULD BE NOTED THAT THE AUXILIARY VARIABLE FEATURE CAN  FIRO3750
C      ALSO BE USED TO DEAL WITH PROBLEMS CONTAINING EQUALITY CON-  FIRO3760
C      STRAINTS.  ANY EQUALITY CONSTRAINT IMPLIES THAT A GIVEN VAR-  FIRO3770
C      IABLE IS NOT TRULY INDEPENDENT.  THEREFORE, IN GENERAL, ONE  FIRO3780
C      VARIABLE INVOLVED IN AN EQUALITY CONSTRAINT CAN BE RENUMBERED  FIRO3790
C      FROM THE SET OF NV INDEPENDENT VARIABLES AND ADDED TO THE SET  FIRO3800
C      OF NAV AUXILIARY VARIABLES.  THIS USUALLY INVOLVES RENUMBERING  FIRO3810
C      THE INDEPENDENT VARIABLES OF THE GIVEN PROBLEM.  FIRO3820
C                                                                    FIRO3830
C      SUBROUTINES AND FUNCTIONS REQUIRED  FIRO3840
C                                                                    FIRO3850

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CC      FUNCTION KE(X)                                FIR02760
C      EVALUATE CONSTRAINTS. SET KE=0 IF NO IMPLICIT CONSTRAINT IS    FIR02770
CC      VIOLATED, OR SET KE=1 IF ANY IMPLICIT CONSTRAINT IS VIOLATED.  FIR02780
C      DIMENSION X(4)                                FIR02790
C      X1 = X(1)                                       FIR02800
C      X2 = X(2)                                       FIR02810
C      KE = 0                                           FIR02820
C      X(3) = X1 + 1.732051*X2                         FIR02830
C      IF (X(3) .LT. 0. .OR. X(3) .GT. 6.) GO TO 1      FIR02840
C      X(4) = X1/1.732051 -X2                         FIR02850
C      IF (X(4) .GE. 0.) RETURN                       FIR02860
CC      1 KE = 1                                       FIR02870
C      RETURN                                          FIR02880
C      END                                             FIR02890
CC      CC -                                           FIR02900
C      FUNCTION FE(X)                                  FIR02910
C      DIMENSION X(4)                                  FIR02920
CC      THIS IS THE OBJECTIVE FUNCTION.                FIR02930
C      FE= -(X(2)**3 *(9.-(X(1)-3.)**2)/(46.76538))    FIR02940
C      RETURN                                          FIR02950
C      END                                             FIR02960
C      METHOD                                           FIR02970
C      THE COMPLEX METHOD IS AN EXTENSION AND ADAPTION OF THE SIM-    FIR02980
C      PLEX METHOD OF LINEAR PROGRAMMING. STARTING WITH ANY ONE      FIR02990
C      FEASIBLE POINT IN N-DIMENSION SPACE A "COMPLEX" OF 2*N      FIR03000
C      VERTICES IS CONSTRUCTED BY SELECTING RANDOM POINTS WITHIN THE FIR03010
C      FEASIBLE REGION. FOR THIS PURPOSE N COORDINATES ARE FIRST    FIR03020
C      RANDOMLY CHOSEN WITHIN THE SPACE BOUNDED BY EXPLICIT CON-    FIR03030
C      STRAINTS. THIS DEFINES A TRIAL INITIAL VERTEX. IT IS THEN    FIR03040
C      CHECKED FOR POSSIBLE VIOLATION OF IMPLICIT CONSTRAINTS. IF   FIR03050
C      ONE OR MORE ARE VIOLATED, THE TRIAL INITIAL VERTEX IS DISPLACED FIR03060
C      HALF OF ITS DISTANCE FROM THE CENTROID OF PREVIOUSLY SELECTED FIR03070
C      INITIAL VERTICES. IF NECESSARY THIS DISPLACEMENT PROCESS IS  FIR03080
C      REPEATED UNTIL THE VERTEX HAS BECOME FEASIBLE. IF THIS FAILS FIR03090
C      TO HAPPEN AFTER 5*N+10 DISPLACEMENTS, THE SOLUTION IS ABAND- FIR03100
C      ONED. AFTER EACH VERTEX IS ADDED TO THE COMPLEX, THE CURRENT  FIR03110
C      CENTROID IS CHECKED FOR FEASIBILITY. IF IT IS INFEASIBLE,    FIR03120
C      THE LAST TRIAL VERTEX IS ABANDONED AND AN EFFORT TO GENERATE  FIR03130
C      AN ALTERNATIVE TRIAL VERTEX IS MADE. IF 5*N+10 VERTICES ARE  FIR03140
C      ABANDONED CONSECUTIVELY, THE SOLUTION IS TERMINATED.         FIR03150
C      IF AN INITIAL COMPLEX IS ESTABLISHED, THE BASIC COMPUTATION  FIR03160
C      LOOP IS INITIATED. THESE INSTRUCTIONS FIND THE CURRENT WORST  FIR03170
C      VERTEX, THAT IS, THE VERTEX WITH THE LARGEST CORRESPONDING    FIR03180
C      VALUE FOR THE OBJECTIVE FUNCTION, AND REPLACE THAT VERTEX BY  FIR03190
C      ITS OVER-REFLECTION THROUGH THE CENTROID OF ALL OTHER VERTICES. FIR03200
C      (IF THE VERTEX TO BE REPLACED IS CONSIDERED AS A VECTOR IN   FIR03210
C      N-SPACE, ITS OVER-REFLECTION IS OPPOSITE IN DIRECTION, IN-   FIR03220
C      CREASED IN LENGTH BY THE FACTOR 1.3, AND COLLINEAR WITH      FIR03230
C      FIR03240
C      FIR03250
C      FIR03260
C      FIR03270
C      FIR03280
C      FIR03290
C      FIR03300

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C          VALUES IF NONE ARE GIVEN, NOT VALUES WHICH ARE MAGNITUDES
C          ABOVE OR BELOW THE EXPECTED SOLUTION.  INPUT VALUES ARE
C          SLIGHTLY ALTERED BY BOXPLX.
C
C          YMN  THIS OUTPUT IS THE VALUE (REAL*4) OF THE OBJECTIVE FUNC-
C          TION, CORRESPONDING TO THE SOLUTION POINT OUTPUT IN XS.
C
C          IER  INTEGER ERROR RETURN.  TO BE INTERROGATED UPON RETURN
C          FROM BOXPLX.  IER WILL BE ONE OF THE FOLLOWING:
C
C          =-1  CANNOT FIND FEASIBLE VERTEX OR FEASIBLE CENTROID
C          AT THE START OR A RESTART (SEE 'METHOD' BELOW).
C          =0   FUNCTION VALUE UNCHANGED FOR 'N' TRIALS.  (WHERE
C          N=6*NV+10) THIS IS THE NORMAL RETURN PARAMETER.
C          =1   CANNOT DEVELOP FEASIBLE VERTEX.
C          =2   CANNOT DEVELOP A NO-LONGER-WORST VERTEX.
C          =3   LIMIT ON TRIALS REACHED.  (NTA EXCEEDED)
C          NOTE:  VALID RESULTS MAY BE RETURNED IN ANY OF THE
C          ABOVE CASES.
C
C          EXAMPLE OF USAGE
C
C          THIS EXAMPLE MINIMIZES THE OBJECTIVE FUNCTION SHOWN IN THE
C          EXTERNAL FUNCTION FE(X).  THERE ARE TWO INDEPENDENT VAR-
C          IABLES X(1) & X(2), AND TWO IMPLICIT CONSTRAINT FUNCTIONS
C          X(3) & X(4) WHICH ARE EVALUATED AS AUXILIARY VARIABLES (SEE
C          EXTERNAL FUNCTION KE(X) ).
C
C          DIMENSION  XS(4),XU(2),XL(2)
C
C          STARTING GUESS
C          XS(1) = 1.0
C          XS(2) = 0.5
C          UPPER LIMITS
C          XU(1) = 6.0
C          XU(2) = 6.0
C          LOWER LIMITS
C          XL(1) = 0.0
C          XL(2) = 0.0
C
C          R = 9./13.
C          NTA = 5000
C          NPR = 50
C          NAV = 2
C          NV = 2
C          IP = 0
C
C          CALL BOXPLX (NV,NAV,NPR,NTA,R,XS,IP,XU,XL,YMN,IER)
C          WRITE(6,1) ((XS(I),I=1,4),YMN,IER)
C          1 FORMAT (////,' THE POINT IS LOCATED AT (XS(1)= ',4(E13.7,5X),
C          1//,' AND THE FUNCTION VALUE IS ',E13.7,' IER = ',15)
C
C          STOP
C          END
C

```

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